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Auditory Source Width perception in Cochelar Implantees and Normal Hearing Individuals

Master's Thesis in the Master's programme in Sound and Vibration

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Department of Civil and Environmental Engineering
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Göteborg, Sweden 2014

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Abstract

Sound source localization is a natural phenomena enabling Normal Hearing listeners(NH) to differentiate and utilize auditory information. A particular case being the cocktail party effect where a NH listener is able to hear the voice of a relevant talker amid a noisy environment. This ability to localize a sound is limited or sometimes non existent in hearing impaired individuals and is most likely to get worse with age. Studies have investigated and concluded that older hearing impaired(HI) individuals have difficulty in sound source localization. However little literature is available on the sound source localization by Cochlear Implantees(CI) and it is known by the experience of CI that they pose a strong challenge in distinguishing between sound sources.

The aim of the thesis is to present Auditory Source Width(ASW) perception thresholds in CI and NH. Auditory Source Width is an attribute of sound which parameterizes sound source perception by an individual. When one is sitting in an opera house listening to a solo by a soprano the voice of the signer is the only source of music. After a moment when a philharmonic orchestra fills the opera house with the symphony what one perceives is no longer a point sound source. Instead orchestra is a sound source which produces an auditory effect perceived to be broader than the soprano alone. CI and NH listeners participated in ASW threshold experiments conducted in a free field audiometry lab comprising a loudspeaker array of 31 speakers. Signals of varying ASW were presented through the loudspeaker array. Subsequently participants were asked to choose the stimuli which they perceived as the broadest of all. The ASW threshold results establish that CI have a very limited ability in perceiving auditory source width.

Keywords: Cochlear Implantees(CI), Normal Hearing(NH), Hearing Impaired(HI), Auditory Source Width(ASW).

Contents

Acknowledgements

The inspiration for the work came along while I was studying at the division of Applied Acoustics, Chalmers University of Technology, Sweden. I got an opportunity to be an Erasmus student from Chalmers and thus I commenced upon the thesis at the Department of Experimental Audiology located at the medical campus of Otto Von Guericke Universität Magdeburg, Germany. Gratitude to both of these institutions and the respective countries which made feel *gemütlich* i.e Sweden and Germany.

An earnest thanks to you Jesko. Apart from your precious supervision, working with you instilled a research attitude in me which I didn't completely posses. Thus now I am ready to pursue my research interests in a more confident way. Special recognition to Wiebke for her unlimited help and for being a delightful office mate.

Credits to Jan, Björn and Michael for their assistance in setting up the laboratory. Appreciation to you Wolfgang. I couldn't have had an examiner as supportive as you. Special acknowledgment to Alice. Your reference for me was the stepping stone of this thesis work.

Jag gillar Sverige och manniskörna är vackra. It was the very first few of lines I could speak in Swedish and it has stayed true every since. Its been more than two years I have been away from home. However the dearness of friends in Sweden and Germany kept me warm enough for the winters here. A vote of thanks to all the seniors at the department of Applied Acoustics and I admire their support.

Friends and family in India, a heartiest thanks. Distance kept me away yet you all have been omnipresent in my conscious to help me thrive well.

1 Introduction

Spatial perception of sound or sound source localization plays a very vital role to enable us perform our daily life activities. Our hearing mechanism is very smart as one can concentrate on the sound one wants to hear and mask out the rest. The classic example being the Cocktail Party Effect where one is able to listen to the target speaker amidst a noisy environment of a cocktail party. Auditory Source Width(ASW) is an attribute of sound which gives an impression of how broad a sound source is in space. It holds importance because it helps us to localize a sound source which further helps to differentiate from more than one sound sources. In their studies [Whitmer, Seeber and Akeroyd 2012] found that older hearing impaired(HI) individuals are insensitive to apparent auditory source width perception. They conducted listening tasks through headphone representation to compare the interaural coherence(IC) thresholds between normal hearing(NH) and HI listeners.

The motivation behind this thesis work is to study auditory source width perception of Cochlear Implantees(CI). It is known by the experience of CI that they have serious difficulty in sound source localization. However no prior work has been done to study the ASW perception of cochlear implantees. The present work aims at measuring ASW thresholds for CI and normal hearing(NH) individuals. It is done in a sound free field environment with the help of a loudspeaker array having 31 speakers. The second chapter presents an introduction to the fundamentals of our hearing system followed by different types of hearing losses. It is lead by a description on various ways of curing the different hearing loss. It also includes an elaborated introduction to how a cochlear implant works. Two relevant pieces of work discussing apparent auditory source width insensitivity in older HI [Whitmer, Seeber and Akeroyd 2012] and localization of sound in rooms [Rakerd and Hartmann 2010] are also discussed in this chapter.

It is of key importance to mention here that most of the literature has investigated apparent auditory source width. However the present works focuses on auditory source width(ASW) which is physically attained by altering the width of the sound source achieved with the help of the loudspeaker array. It is in contrast with the apparent auditory source width focused in the literature.

In different works apparent auditory source width has been fabricated by superimposing multiple sound reflections or by varying the interaural coherence of a sound. In the thesis on all occasions ASW has been referred to auditory source width except in the literature chapter where it is explicitly mentioned.

The third chapter describes the experimental setup i.e. the free field audiometry room, the loudspeaker array and more importantly the calibration of the loudspeakers. The next chapter illustrates the generation of sound stimuli with different auditory source width and how they are presented through an array of 31 loudspeakers.

The fifth chapter presents the relation between interaural time differences(ITD), Interaural Coherence(IC) and auditory source width(ASW). An artificial head was used to produce binaural recordings and measure the ITD's. These are finally used to calculate IC values and establish a relation with ASW. In sixth chapter the methodology used to conduct the experiments is explained. Alternate forced choice procedure(AFC) is used to implement the experiments. In this method a participant is presented with multiple stimuli and eventually the participant has to choose one of the stimuli. Chapter seven presents the results from ASW thresholds experiments for normal hearing individuals and cochlear implantees. It includes comments on the independent participant and mean collective response.

The last two chapters consists of a discussion and a conclusion of the thesis work. While the former chapter compares the results of the NH individuals with that of the work from [Whitmer, Seeber and Akeroyd 2012] whereas the later concludes the thesis work with the possible applications and future scope of work.

2 Basic Aspects of Hearing

2.1 Hearing and Cochlea

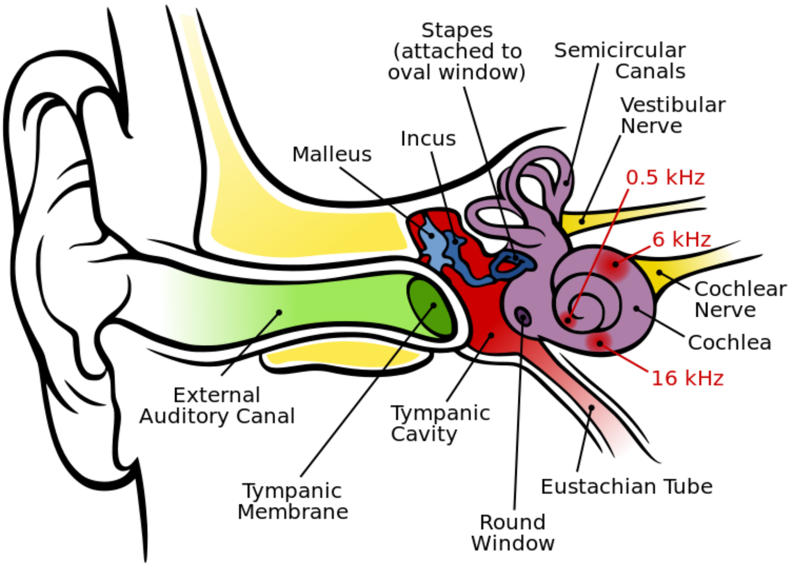


Figure 2.1: Parts of a human ear and frequency regions in a cochlea, ref: [Wikimedia Commons Anatomy of Human Ear]

Cochlea is the final interface in the human hearing system which transmits the sound signals to the brain. The incident sound waves travels through the External Auditory Canal and strikes the tympanic membrane. It instigates a Malleus and Incus imitating a *hammer strike action* which further excites the Cochlea through the Oval Window. The Cochlea consists of a basilar membrane. Different regions of the membrane are activated according to the frequency content of the sound signal as they are shown in figure 2.1. This excitement of the basilar membrane finally transmits signal through the cochlear nerve and at last interpreted in the brain.

2.2 Hearing Loss

Hearing losses can be mainly classified as-

- Sensorineural Hearing Loss(SNHL)- This type of hearing loss can be attributed to malfunctioning of the following three-
  The vestibulcochlear nerve, the inner ear or the auditory processing in the brain. The

vestibularcochlear nerve which carries the name cranial nerve VIII in anatomical terminology is shown in figure 2.2. It is responsible for transmitting sound and body balance information from the brain to the inner ear.

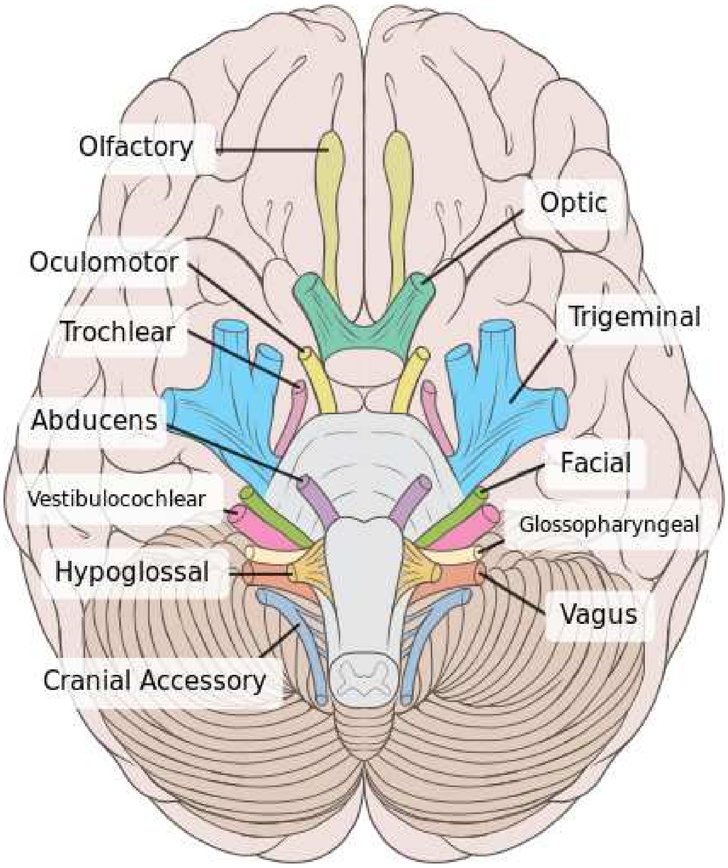


Figure 2.2: A section of a human brain depicting majority of nerves responsible for controlling different body functions, ref:[Wikimedia Commons Human Head]

In many of the cases of SNHL damaged hair cells of the organ of Corti is the dominant reason for the hearing loss. These hair cells are present around the basilar membrane and a specific bunch of them gets excited once an incoming sound activates a particular section of basilar membrane. These are shown in figure 2.3. This type of damage is irreparable. It may be prevalent since birth or may occur due to prolonged exposure to high levels of sound.

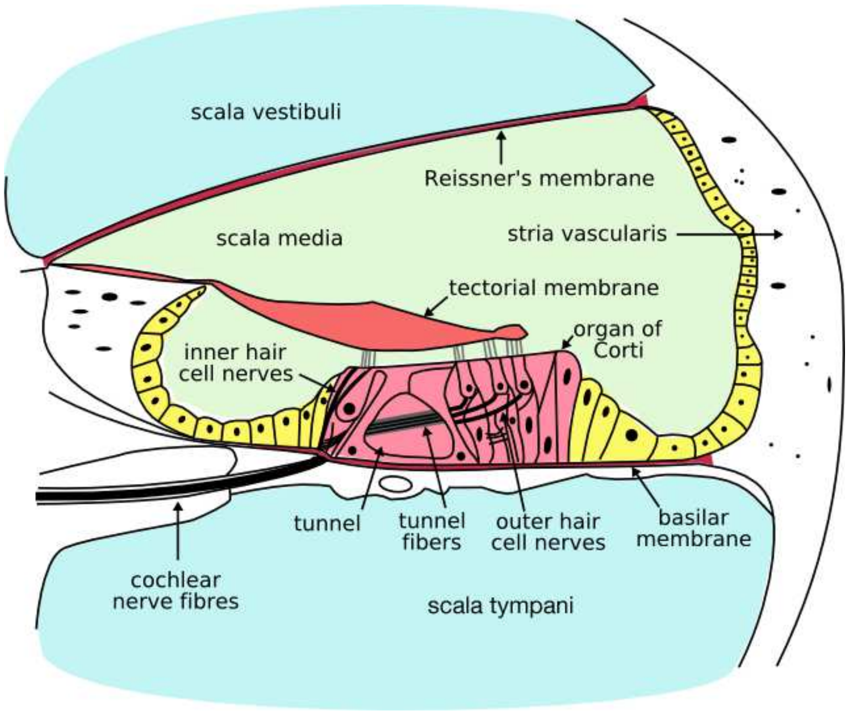


Figure 2.3: A Cochlear Cross-section displaying hair cells around the basilar membrane, ref: [Wikimedia Commons Cross-Section of Cochlea]

- Conductive Hearing Loss- It occurs when sound is not completely transferred from the outer ear through the auditory canal to the Tympanic Membrane. This may happen due to multiple reasons. Two of the common reasons are- Fluid or wax formation in the Eustachian tube due to an infection or tumor. These at some instances can be treated by surgery. Perforations in the Tympanic Membrane is another reason for this type of hearing loss.

- Mixed Type- In many of the cases a person may develop an hearing impairment both due to SNHL and Conductive Hearing Loss.

2.3 Hearing Aids

The different types of hearing aids are classified according to the type of hearing loss they treat. The three major types are discussed in brief below-

- Behind The Ear (BTE)- It is one of the most common type of hearing aids. It can be used to treat mild to severe hearing loss. It consists of a casing and an ear mold connected to each other. The case rests behind the Pinna while the mold is inserted at the entrance of the ear canal. The casing consists of a sound receiver, processor and battery. The incident sound waves are amplified by the amplifier. Upon amplification they are emitted by the ear mold in the ear canal for an improved hearing.

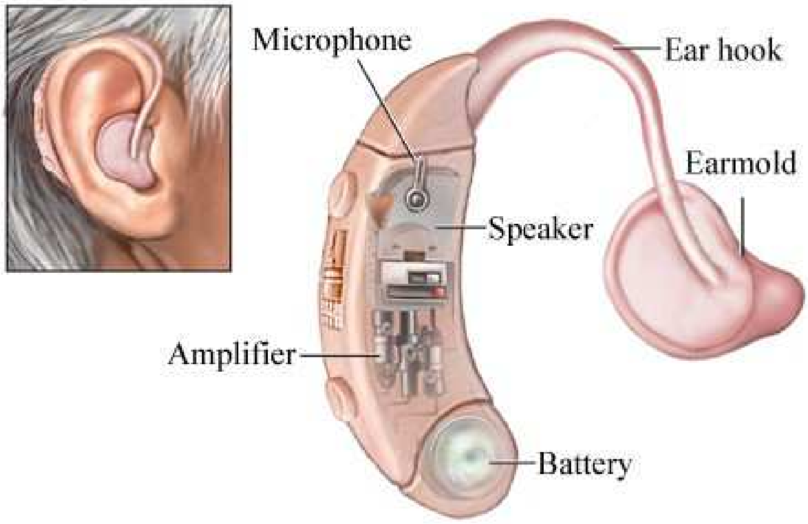


Figure 2.4: Different parts of a behind the ear hearing aid, ref: [Behind The Ear Implant]

- Bone Conduction Hearing Aid - This type of hearing aid works on the principle of bone conduction. It is the conduction of sound to the inner ear by the skull bone. In this case the middle ear is bypassed and sound is directly transmitted to the inner ear. An electromagnetic transducer converts the sound signals in to mechanical vibrations and transmits these vibrations to the inner ear through the cranial bones. It is used in the cases of conductive type of hearing loss. These further can be classified in two sub types and they are as followed.

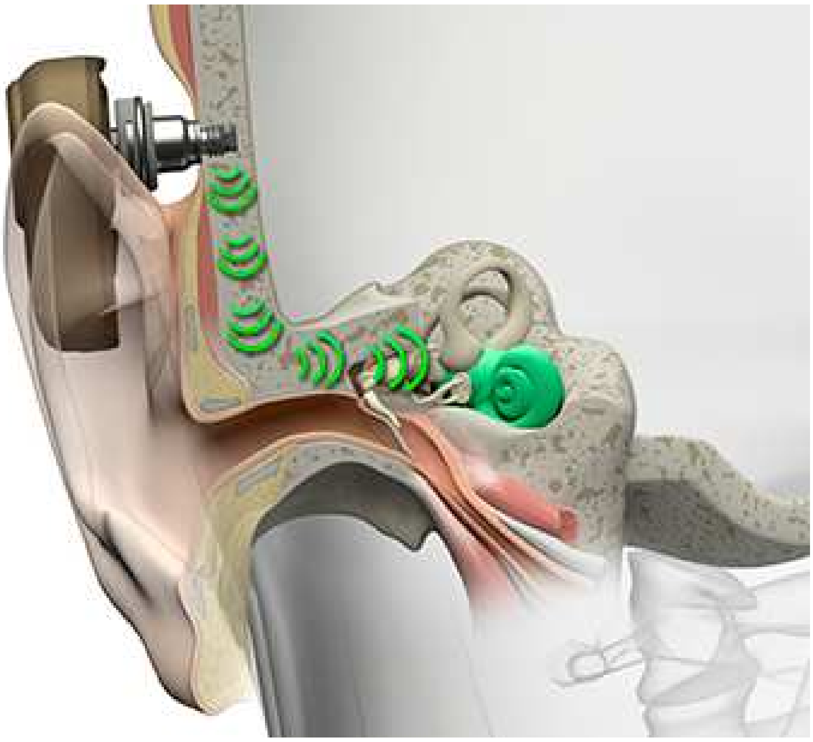


Figure 2.5: Bone Anchored Hearing Aid consisting of an abutment on the outside screwed to a piezoelectric material present under the skin, ref:[BAHA Chalmers University of Technology]

- Bone Anchored Hearing Aid (BAHA)- It consists of a sound processor on the outside of the head called an abutment which is screwed to a titanium prosthesis placed inside the skull as shown in figure 2.5 . This abutment contains contains a piezoelectric transducer. It transmits the vibrations to the inner ear through the skull. This finally excites the cochlea and thus the signal is transmitted to the brain.

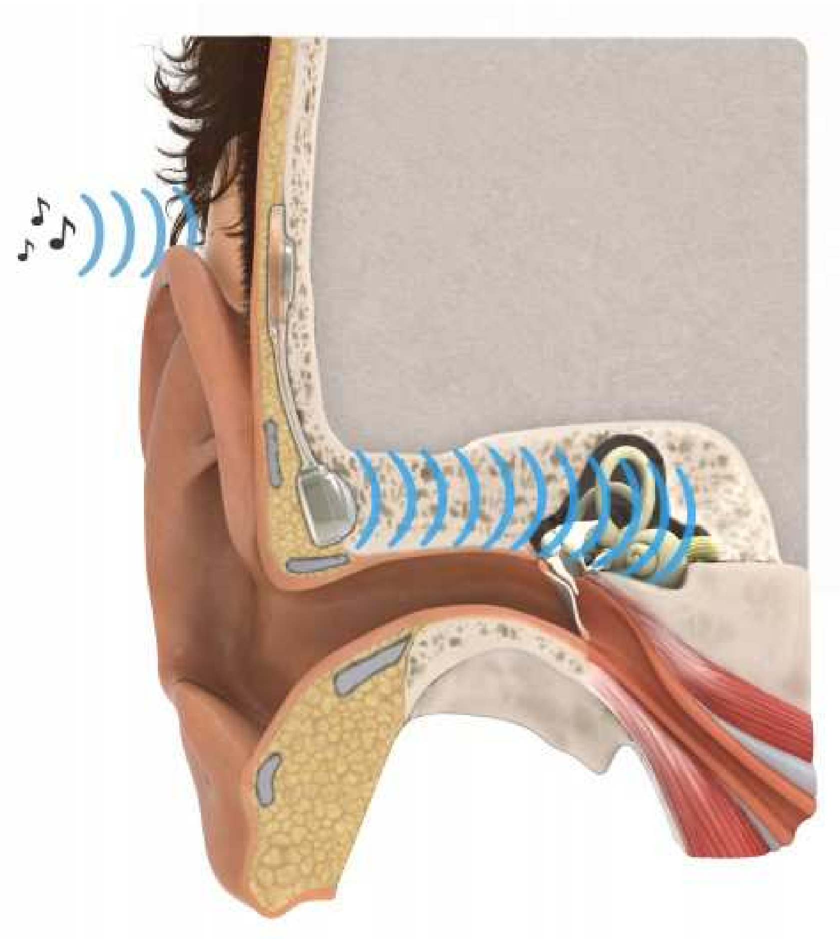


Figure 2.6: Bone Conduction Implant consisting a magnetically coupled processor on the outside to the implant under the skin, ref: [BCI,Chalmers University Of Technology]

- Bone Conduction Implant (BCI)- It was developed in 2012 in collaboration with Chalmers University Technology and Sahlgrenska City Hospital in the city of Göteborg, Sweden. Unlike BAHA this is free from any form of screw. It consists of two major parts. A processor and an implant. The implant is inserted in the bone behind the ear, figure 2.6. It is magnetically attached to the processor present outside. The other end of the implant which is a transducer vibrates and emits signals to excite the cochlea.

2.3.1 Cochlear Implants

Severe to profound SNHL can be treated by implanting a person with cochlear prosthesis. A simple model of the implant is presented in figure 2.7 and described thereafter.

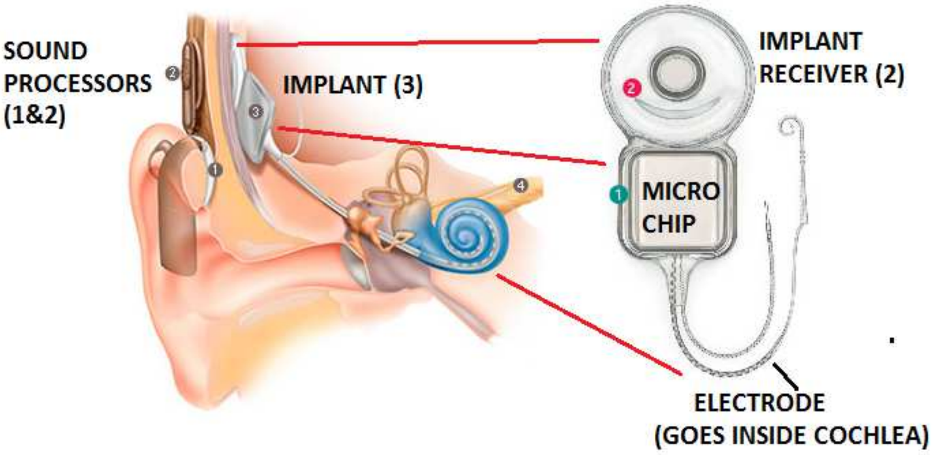


Figure 2.7: Overview of a cochlea implant, ref: [Cochlear Implant]

A microphone present on the outside receives the incoming sound waves which are processed and converted into electrical signals by the speech processor. These processed signals are then transmitted to a receiver present beneath the skin. The receiver extends as an array of electrodes on the other end which travel up to the inner ear. These electrodes are planted in such a way that they penetrate the cochlea and take up its spiral shape. There are two coils each on the outside and inside of the head as presented in figure 2.8 namely the Transmitter and the Receiver coils. They can interact either through the Transcutaneous or through Percutaneous communication. In former type the communication takes place with help of radio frequencies. There is a magnet in the implant which not only holds together the Transmitter and the Implant but it is also responsible for the transfer of signals through electromagnetic induction. Majority of the cochlear implants available today work on Transcutaneous communication. In case of Percutaneous interaction there is a physical link, piercing through the skin, between the Transmitter and the Receiver.

Figure 2.8: Cochlea Implant Parts. The image on the left and right side are the respective parts which are placed on the outside and inside of the ear, ref: [Parts Of Cochlear Implant]

A more detailed description of the functioning of the implant is presented hereupon. Cochlea consists of Basilar Membrane which acts as a bandpass filter and divides the incoming sound into different frequency components. The foremost part of the membrane is the widest part and is called as Base. Thereupon membrane becomes thinner and thinner as it forms a spiral and terminates at the Apex. Each region on the membrane is sensitive to a particular center frequency as shown in figure 2.9.

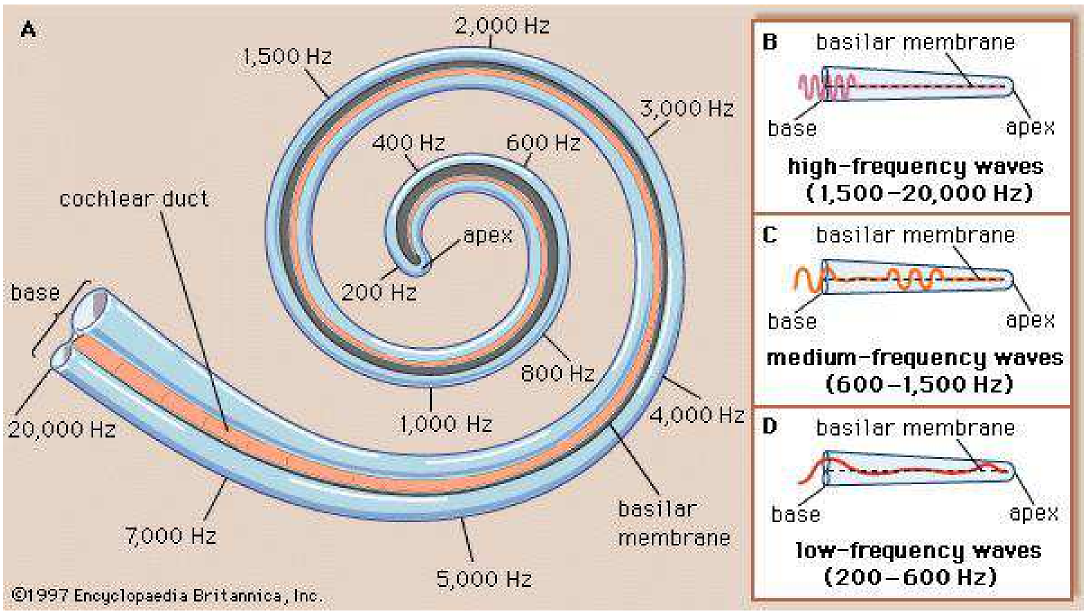


Figure 2.9: An image of basilar membrane with different frequency regions , ref [Basiliar Membrane Britannica]

Next comes the role of hair cells. Hair cells are present around the basilar membrane. In case of normal hearing individuals each of the frequency regions on the membrane are responsible for activating a specific bunch of hair cells. These cells further release an electrochemical substance which fires neurons and they finally transmit the signal to the brain through the central nervous system. However severe to profound SNHL is associated with damage of these hair cells and unfortunately it is an irreparable damage.

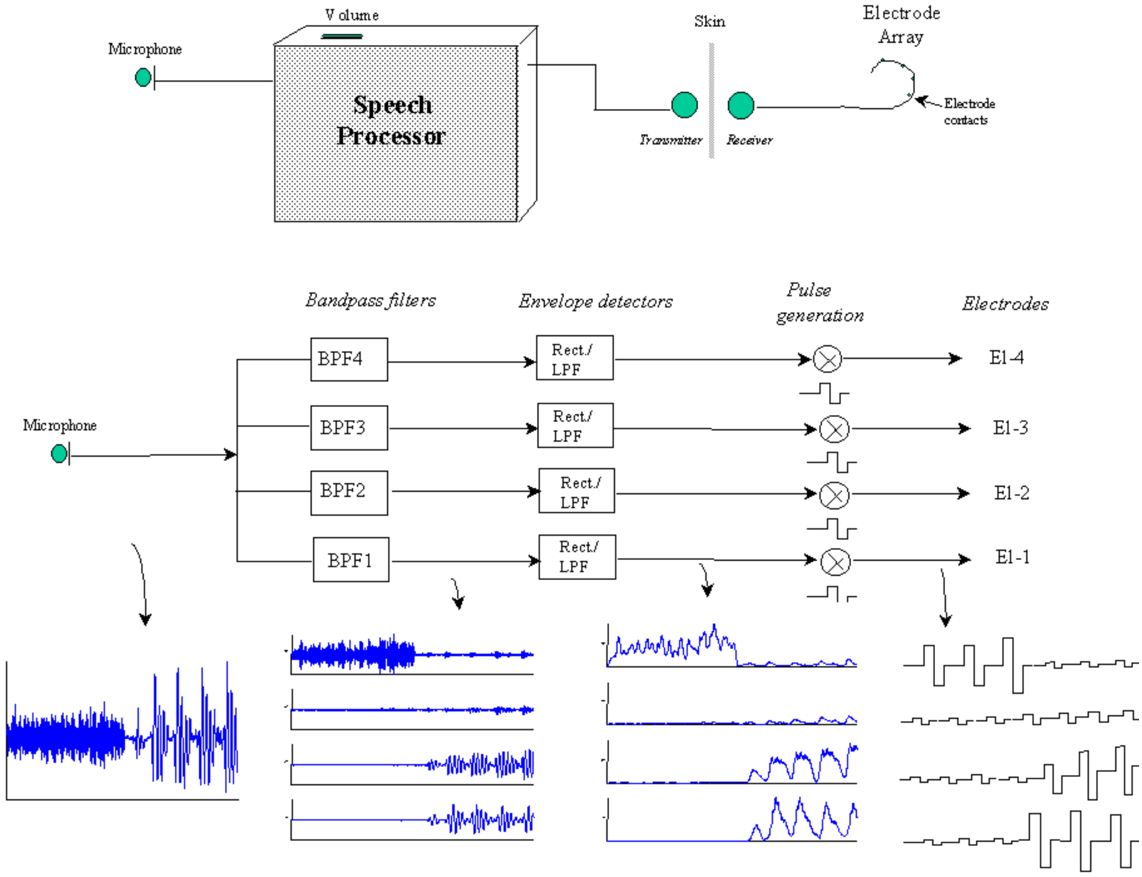


Figure 2.10: Functions of different parts of a cochlear implant, ref:[Philipos C. Loizou]

Figure 2.10 presents a functional view of different stages of cochlear implant. After the signal has been received by the microphone the speech processor divides the signal in different frequency bands quiet similar to how the division of frequency is present on the basilar membrane. After the frequency based signal division comes the envelope extraction.

A sound specifically speech can be divided into temporal fine structure(TFS) and an envelope(E). An illustration is presented in figure 2.11. In simplicity TFS are the rapid variations in time in the signal which is also responsible for the dynamic nature of the signal. In contrast to the fine structure E refers to the slow changes. In a research [Brian C.J Moore 2008] studied the importance of TFS in order to perceive sounds in the presence of masking sounds. It can be said that in the situation where only a single sound source is present the E cues are enough for communication. However in complex situations having more than one talker it is vital to have TFS cues in addition to the E cues. In the beginning of research CI used to extract only E cues until recently when one of the major implant manufacturer MEDEL introduced the sound processor SONNET [SONNET MEDEL] which uses TFS cues. After this extraction signals are generated in the form of pulses in the implant which sits underneath the skin coupled with the external sound processor. These pulses are fed into respective electrodes. These electrodes are further inserted in the cochlear and eventually they are responsible for firing the auditory neurons upon arrival of a pulse.

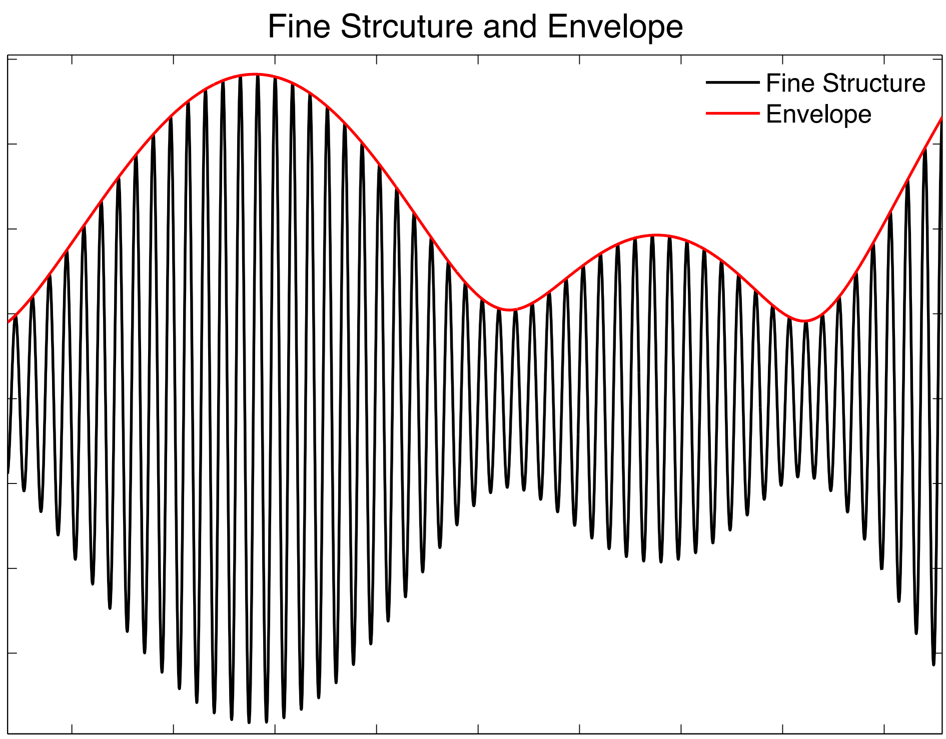


Figure 2.11: Envelope and Fine Structure of sound signal

2.4 Apparent Auditory Source Width and Interaural Cross Correlation(IACC)

The study of spatial perception of sound rests mainly on two fields of acoustics i.e. psychoacoustics and architectural acoustics. Former is the study of our mental, emotional and physiological responses to a sound. While later emphasizes on characteristics of the space around us with which the sound interacts before it finally reaches our ear. Auditory spatial impression contains a lot of information about the space around us. It has been further branched in to size impression, spaciousness and reverberance by [Potter 1993] in his work on binaural modeling in rooms. Figure 2.12 presents the overall view of spatial impression.

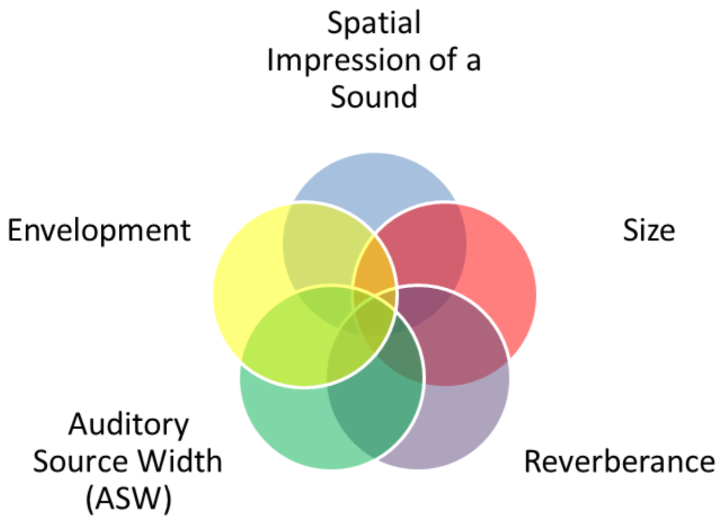


Figure 2.12: Schematic representation of Spatial Impression

Size gives an idea on the structure of the space around us and may be sufficient to reveal some details of the interiors of the space. Next is the reverberance which arises due to the energy contained in the late reflections of the sound.

Spaciousness is defined as "subjective broadening of a sound sources, in the sense that they seem to fill a larger amount of space than is defined by the visual contours of sound source".

Acousticians all around the world have expressed that studying apparent auditory source width of performance halls i.e. Opera houses and Concert halls is a smarter way to analyze the quality and suitability of a particular space. Apparent ASW "*is the apparent auditory source width of the sound field created by a performance entity as perceived by the listener in the audience area of a concert hall.* " However ASW has been related to Interaural Cross-Correlation Coefficient(IACC) by many of the researchers in the past few decades. One of such work has been by [Okano, Beranek and Hidaka 1998]. They performed some experiments with NH individuals and established that [1-$(IACC)_{E3}$] together with sound Strength Index (G_{low}) is one of the best way to describe apparent ASW. Strength index is a measure to identify the relative loudness of a sound source.

IACC tells us about the difference in the sound reaching our both ears. $(IACC)_{E3}$ means the number obtained by averaging the early interaural cross correlation function $(IACC)_E$ in three octave bands with mid frequencies at 500, 1k, 2k Hz. This is the frequency region where wavelength of sound is comparable or smaller than the acoustical distance between the two sides of head. ASW is basically calculated from the sound field generated within the first 80ms of the direct sound. However apart from its relation with the sound field ASW alters with the change in frequency band. Both [Okano, Beranek and Hidaka 1998] and [Sato and Ando 2002] have proved by their separate experiments that ASW of a sound is perceived as broader when the sound contains more lower frequency content.

In the next section literature on the apparent auditory source width perception by the hearing impaired and normal hearing has been presented.

2.5 Literature

The research on spatial perception of Cochlear Implantees(CI) is yet to have a fully developed study on how the CI perceive auditory source width(ASW). However literature is available on the apparent ASW perception by hearing-impaired individuals. One such study has been carried out by William M. Whitmer, Bernhard U. Seeber and Michael A. Akeroyd [Whitmer, Seeber and Akeroyd 2012]. It presented a comparison of the apparent ASW perception between the older HI and normal-hearing(NH) individuals.

The work is based on the hypothesis that older HI individuals are insensitive in horizontal localization of sound. In order to test the hypothesis they conducted headphone based experiments and presented sounds with variable interaural coherence(IC), a cue most associated with apparent ASW. Interaural coherence is a measure to identify the difference in the sound reaching our both ears. This was done to determine the problems HI people have to punctuate sound sources. The punctuateness of the sound can be described as the extant to which a sound is perceived to be present in the left or right in the surroundings. In fact they found that the HI participants perceived the sound source rather diffused or vague in space.

Another separate study [Wiggins and Seeber 2011] discussed the affect of dynamic-range compression of signals on the sound localization. In this research authors found that the lateralization of a sound is affected by the compression of sound in audio devices. It includes normal hearing people and bilateral cochlear implant users. For example the sound with abrupt on-set or off-set were perceived to be more centrally localized. While the sounds with gradual on-set or off-set had more chances of being localized as coming from left or right.
It has also been of the view that the broadness of the sound source increases as the IC is decreased. In their experiments [Whitmer, Seeber and Akeroyd 2012] varied the IC in the headphone representation of the sound. Following are the excerpts from the experiments.

2.5.1 Apparent ASW discrimination of broadband noise based on IC by normal-hearing and hearing-impaired individuals: Experiment 1

In this experiment individuals of difference in age and hearing loss participated. The stimuli were broadband noises corresponding to reference ICs which varied over different stimuli. In one of the earliest works on Interaural Coherence [Pollack and Trittipoe 1959] studied the response of normal-hearing individuals to different interaural correlations. In their work [Whitmer, Seeber and Akeroyd 2012] compared their results for NH listeners to the results from [Pollack and Trittipoe 1959] and observed the two studies has similar results while measuring IC thresholds.

First of all the Pure-tone audiometric threshold of the participants were measured. This attribute provides information on the type and degree of hearing loss of an individual and is presented in figure 2.13. Four of the older participants had variable pure tone threshold of VPTA greater than 20 dB hearing loss(HL). Hence it can be said that the hearing loss in the participants varied over a wide range.

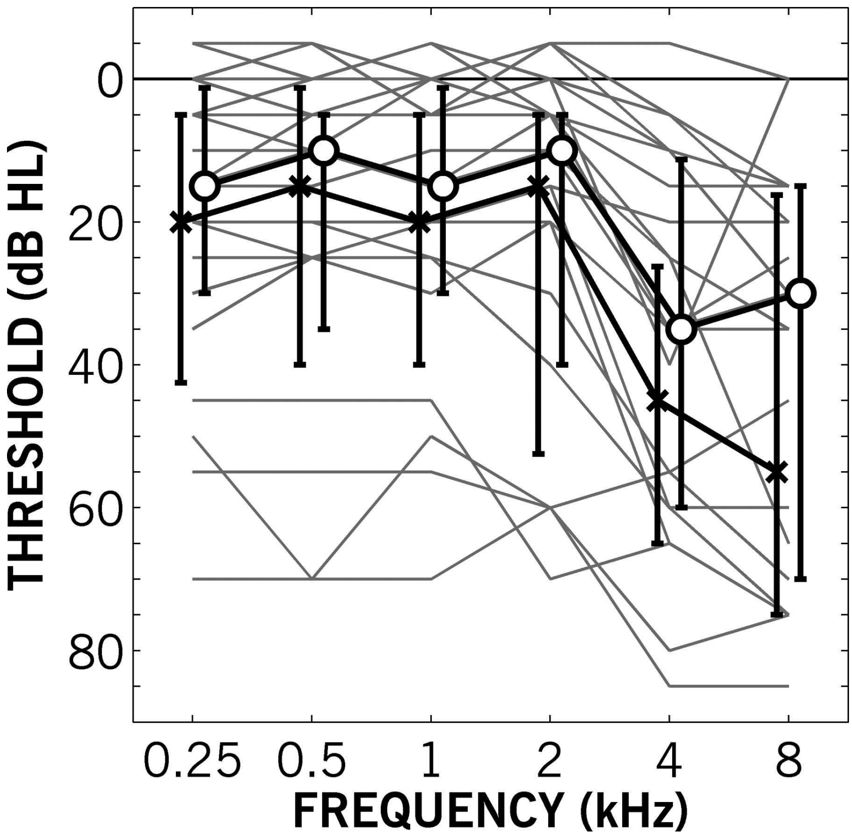


Figure 2.13: Pure-tone audiometric threshold as a function of frequency for Experiment 1. Gray lines show an individual participant's better-ear(based on variable pure-tone threshold average) audiogram. Black lines show median thresholds for left(crosses) and right(circles) ears. Error bar show first and third quartile ranges

In order to produce stimuli of variable ICs uncorrelated narrow band octave noises centered at 250-4000 Hz were used. They were then reproduced using the symmetric generator method [Har and Cho 11]. These signals were further modified to obtain an interaural time difference Δ ITD values of -312, 0 or 312 μs. Two interval forced-choice adaptive procedure was used for the discrimination tasks. Participants were asked to judge which of the two sounds presented appeared wider. The results are presented in figure 2.14

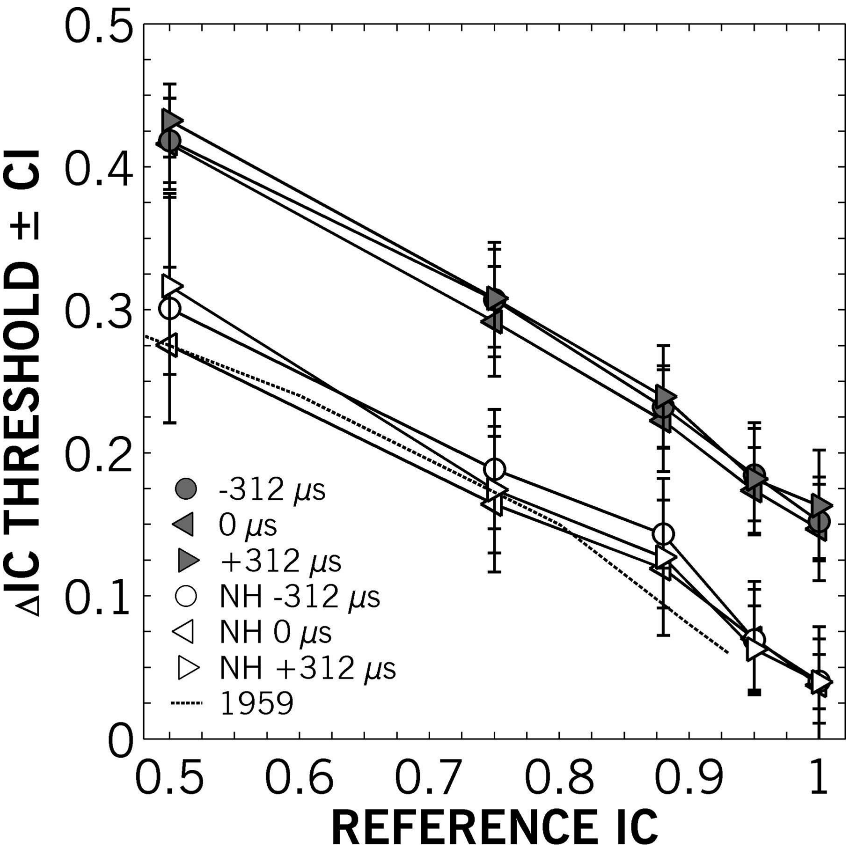


Figure 2.14: Mean IC difference (ΔIC) thresholds as a function of reference IC for older (filled) and younger (open) participant at global ITDs of -312(left triangle), 0(circle) and +312(right triangle)μs. The dashed line is the average data from Pollack and Trittipoe (1959).

In figure 2.14 mean ΔIC thresholds as a function of reference coherence and ITD are presented. These are measured for both age groups i.e. old and young. It is observed here that the thresholds for older groups are higher or worse than the younger group. On observing the responses on the basis of the different ITD values it can be inferred that the thresholds of the either of age group is not significantly affected by the change in the ITD of the signal.
The results from this experiment supports their hypothesis that the older individuals tend to poorly identify the wideness of a sound.

2.5.2 Apparent ASW discrimination of broadband noise based on IC by normal-hearing and hearing-impaired individuals: Experiment 2

In this experiment the participants were asked to draw the visual representation of the signal. They were informed to draw size and position of the sound on a touch screen. Five simulated positions were chosen for the sounds. These were 0 °, ±30° and monoraural left(L) and right(R). They produced ±30° positions using the ITD and ILD values derived from average measurements of the KEMAR head. The monoauraly left and right positions were produced by fully attenuating the opposite channel. IC values of 0.6, 0.8 and 1 were used to test the sounds. The

participants were asked to draw the perceived width of the image. They were provided with an image of a square head. After the presentation of each sound they drew their perception on the square head image. The results from the experiment are presented in figure 2.15 and 2.16. In the following figures grayness indicate the most drawn portions of the image. On comparing the figures 2.15 and 2.16 it can be concluded that NH individuals are precise in punctuating sound images. On comparing the IC values of 0.6 to 1 at $\pm 0°$ the representation by NH participant changes from a strip along the horizontal diameter to a small circle very close to the center. For the same stimuli the HI participant's response is more or less like a exploded sphere at the center.
Thus these results also prove the hypothesis and it can be concluded that the HI people have a great difficulty in perceiving the width of a sound.

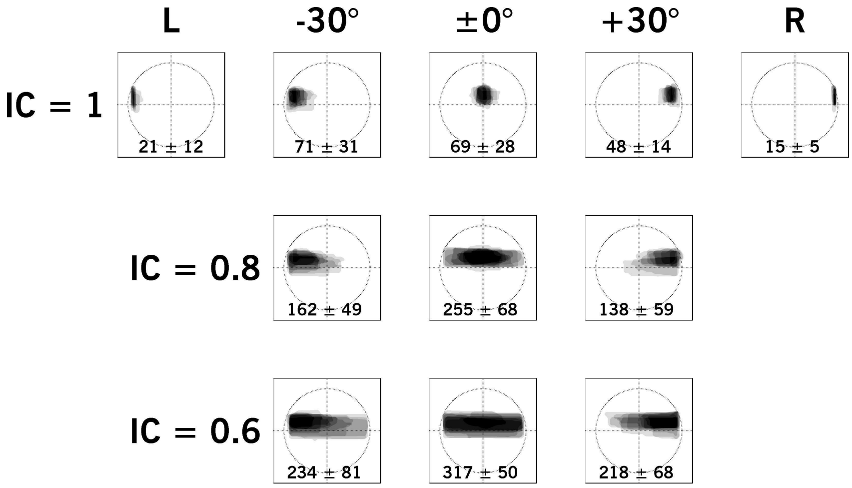


Figure 2.15: Example of the results from a younger NH participant, aged 38 years, BEA(Better ear variable pure tone threshold average) of 6.7 dB HL(hearing loss) showing aggregated images as function of position(horizontal labels). Levels of gray indicate the frequency of response for that pixel.

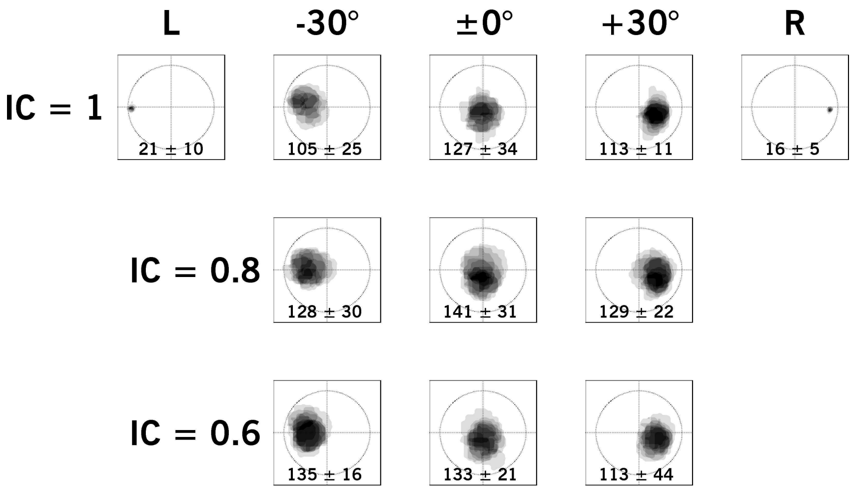


Figure 2.16: Example of the results from an older HI participant, aged 68 years, BEA(Better ear variable pure tone threshold average) of 48.3 dB HL(hearing loss) showing aggregated images as function of position(horizontal labels). Levels of gray indicate the frequency of response for that pixel.

2.5.3 Localization of sound in rooms. V. Binaural coherence and human sensitivity to interaural time differences(ITD) in noise

Interaural time difference(ITD) have an important role in the spatial perception of sound as it effects interaural coherence. ITD is experienced because a sound coming from a direction reaches one ear first and takes time to travel to the other ear. A listener's judgment to localize a sound in space depends upon its ability to discriminate the sounds with different ICs. Hence it was thought to be of importance to study the effect of ITDs on ICs. In a series of experiments [Rakerd and Hartmann 2010] studied localization of sound in rooms. In one of their study they established that ICs vary linearly with change in ITDs. They conducted two set of experiments in this study. In the first experiment they studied change in ICs with different ITDs in different frequency bands. While in the second they observed how the relation between IC and ITD varies in different rooms. The two experiments are discussed in detail hereupon.

2.5.4 Experiment 1: Frequency Band Dependency

In this experiment a set of recordings corresponding to coherence values increasing unit wise from 0.2 to 0.8 were used. These recordings were constructed using a KEMAR manikin and are presented in figure 2.17.

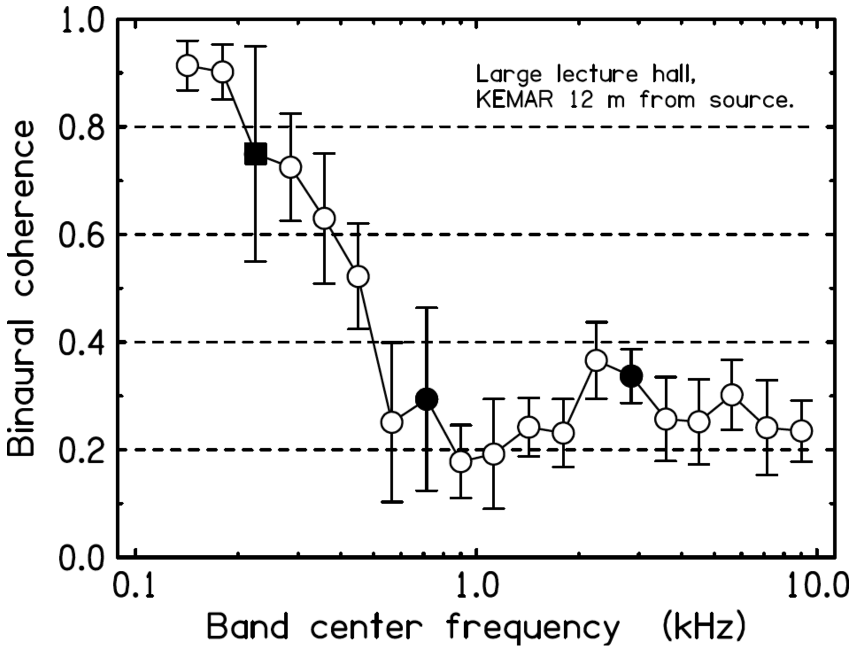


Figure 2.17: Binaural coherence measured in a lecture hall for 1/3 octave noise bands. Measurements were made at 10 different locations within the room with sound source(loudspeaker) and receiver(KEMAR) separated by 12m. Filled circles indicate the "mid and "high" bands

The sounds were then presented to participants via headphones in two runs. First they were presented with the recordings and then with some added time delays. In order to introduce the time delays on the existing ITDs, in some of the trials signal was delayed on the left side and in the rest on the right side. The listeners were asked to mark whether the image moved left or to the right. The results for the mid-frequency bands are presented in figure 2.18. It is evident that with the increase in coherence the percentage consistency in response increases. Also the increase in percentage consistency increased very rapidly after 50μs and specifically for coherence of 0.8 it appears to have almost a hundred percentage consistency.

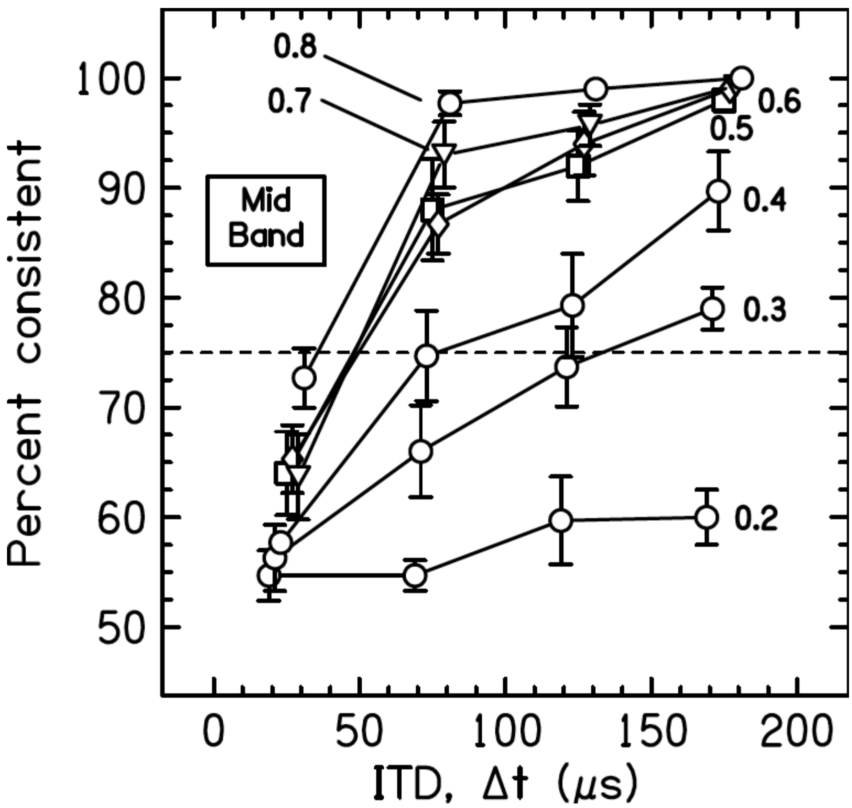


Figure 2.18: Results of Experiment 1 for the mid-frequency noise band(715 Hz). The percentage of lateralization judgements consistent with the sign of the applied interaural time difference(ITD), averaged over the five listeners, is plotted as a function of the ITD with noise band coherence as a parameter.

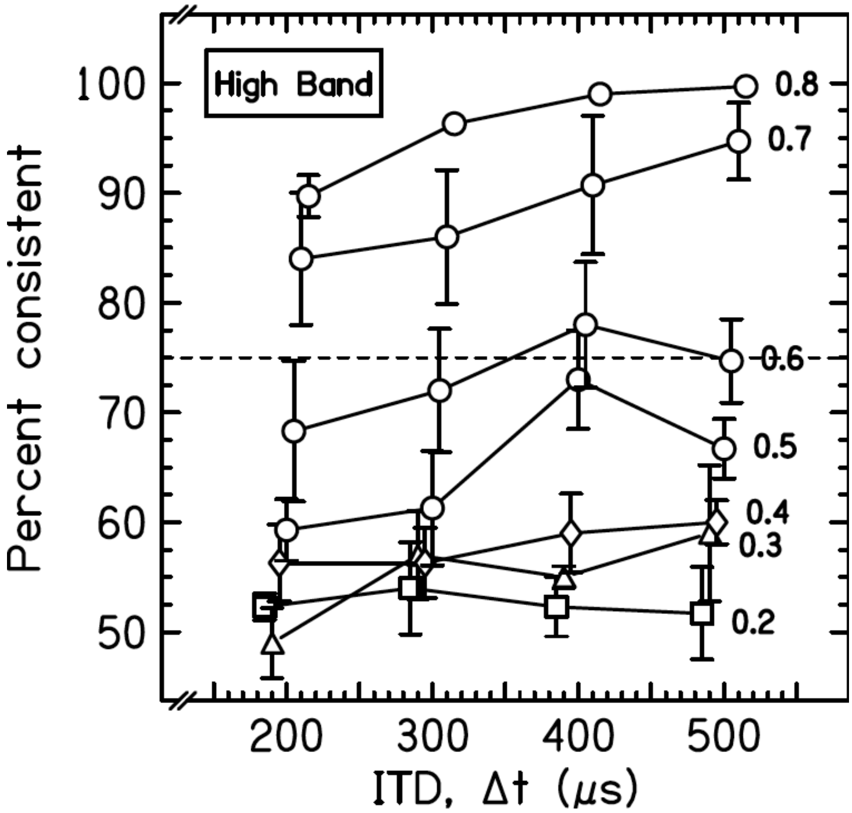


Figure 2.19: Results similar as figure 2.7 but instead for high-frequency band (2850 Hz).

The same results for a high-frequency band are shown in figure 2.18. It is observed that for a coherence of 0.2 the listeners are not able to perform the task and perceive the change on the basis of ITD's. At the same time for the coherence of 0.7 and 0.8 the percentage consistency is relatively high as compared to the lower values of coherence. The time scale for the high-band measurements is longer than for the mid-band frequencies. The reason being the smaller wavelength of sound at higher frequencies needs more time to travel from one ear to the other. In a nutshell it can be said that the percentage consistency increases with the increase in coherence and this increase is more rapid in case of high frequency band.

2.5.5 Experiment 2: Different rooms

It was initially considered that the relation between IC and ITD is also effected by the room in which the listener is in. Primarily it was thought that the reverberation time of the room is important while studying the ITDs. In their subsequent experiment [Rakerd and Hartmann 2010] repeated the above measurements but in three different rooms. The three rooms were-

- Room 1- A classroom with sufficient absorption and a reverberation time of 0.4s in the frequency range of interest.
- Room 2- A laboratory with a reverberation time of 0.8s.
- Room 3- A reverberation room with high diffusion and a varying reverberation time of 1.2s at lower frequencies to 2.5s at higher frequencies.

The results are shown in figure 2.20. It is inferred here that the coherence value of 0.8 has the highest percentage consistency irrespective of the room they are measured in. It can also be deduced that both for the mid-band and the high-band frequencies the percentage coherence increased almost in a similar fashion for all the rooms and it increases rapidly for mid-band than for the high-band values.
Hence different rooms do not substantially effect the interaural coherence. Since these experiments were done in headphone representation therefore it will be interesting to find if interarual coherence varies in a similar way as presented in a sound field and how CI react to the sounds.

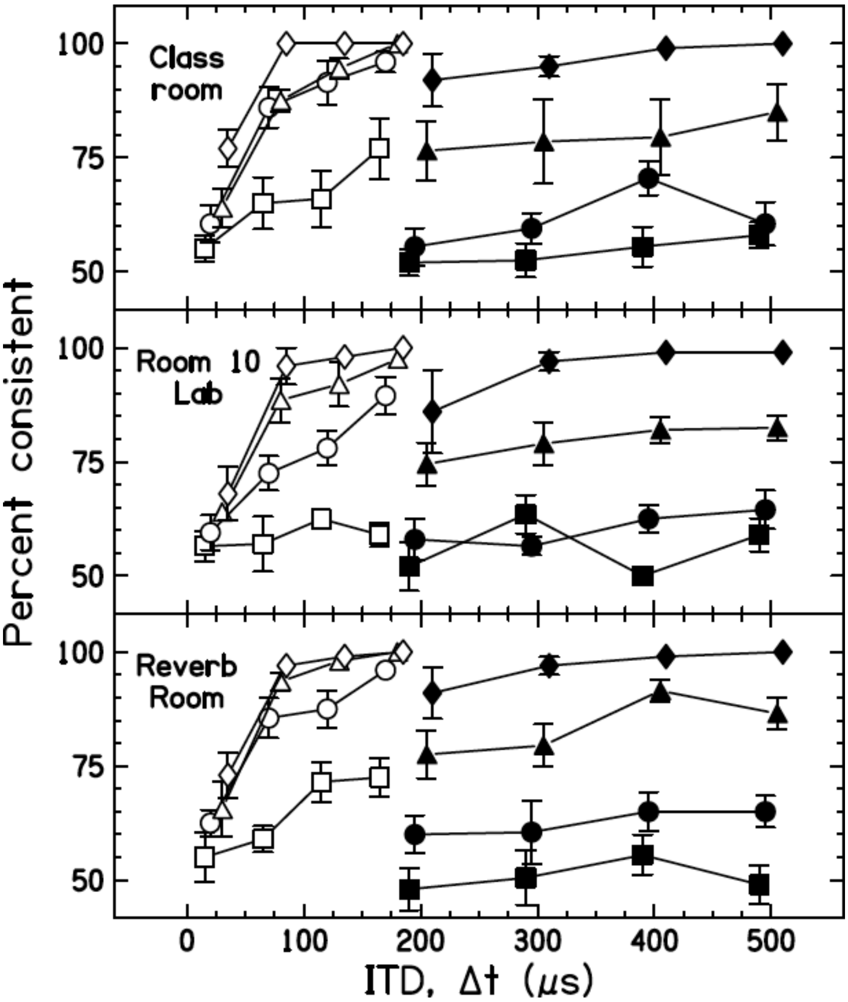


Figure 2.20: Results of Experiment 1 for the mid-band(open symbol) and the high-band(filled symbol) separated out by room. Coherence value for each symbol is $\triangle = 0.65, \square = 0.25, \bigcirc = 0.45, \diamond = 0.8$.

3 Experimental Setup and Calibration

3.1 The setup

The experiments were conducted in the Freefield audiometry room located at the department of Experimental Audiology, Universitätklinikum Magdeburg. The room is cubical in shape. The two side walls and one of the back wall of the room have acoustically treated slabs. There are large windows in one of the walls and the ceiling is made of microscopic absorbers. An array of 31 loudspeakers as shown in the figure 3.1 is used to create the sound free field condition. Genelac 6010A speakers are used for sound generation.

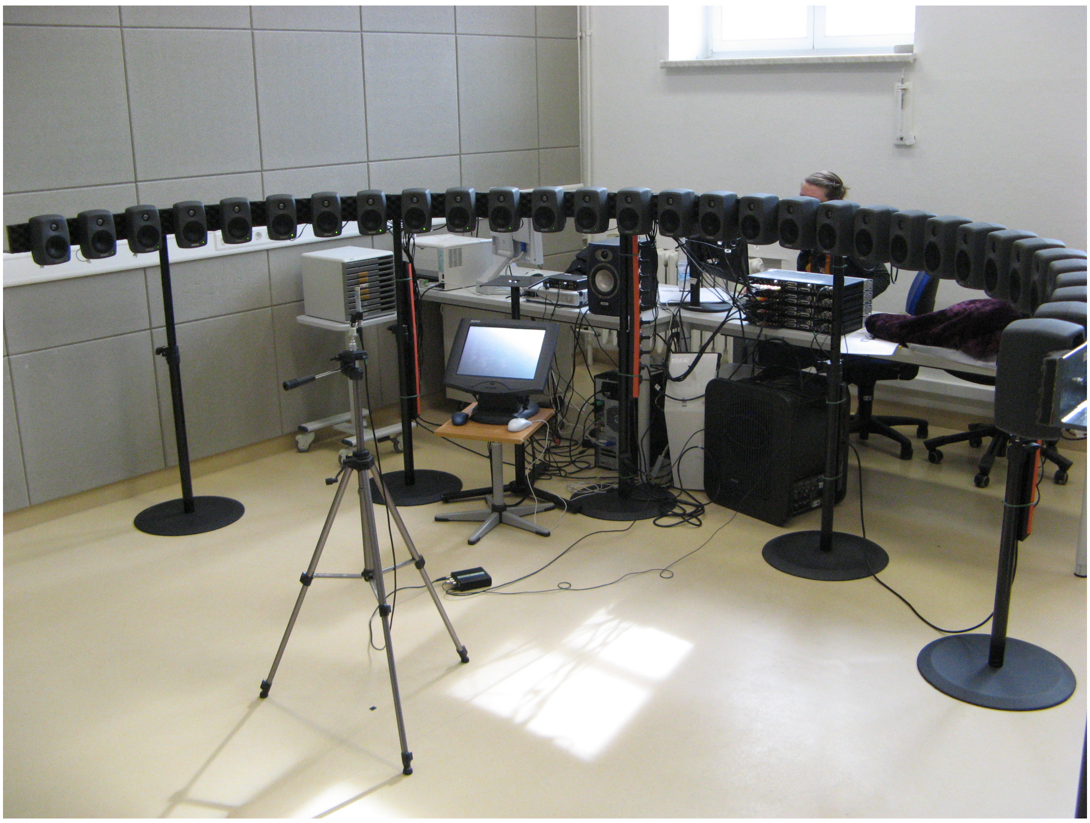


Figure 3.1: Free Field Audiometry Room

In order to achieve multi channel sound reproduction, which is a limitation within windows operating system, Playrec utility was used. It is an open source based platform which gives access to sound cards through PortAudio which is again an open source audio I/O library. RME make ADI-8 QS AD/DA converters were used to produce the signals through the loudspeaker array. The diameter of the ring is 3.5 meters. The loudspeaker center is 1.17 meters above the floor. The floor is observed to be sound reflective. A set of four RME make AD/DA converters are used to feed signal into the array from the PC. A microphone placed at the center of the ring was used to calibrate the array.

3.2 Calibration

The steps taken to calibrate the array are presented in a flowchart on the next page. The frequency response of each loudspeaker was measured by playing a signal containing Gaussian noise through them. The response is presented in figure 3.2.

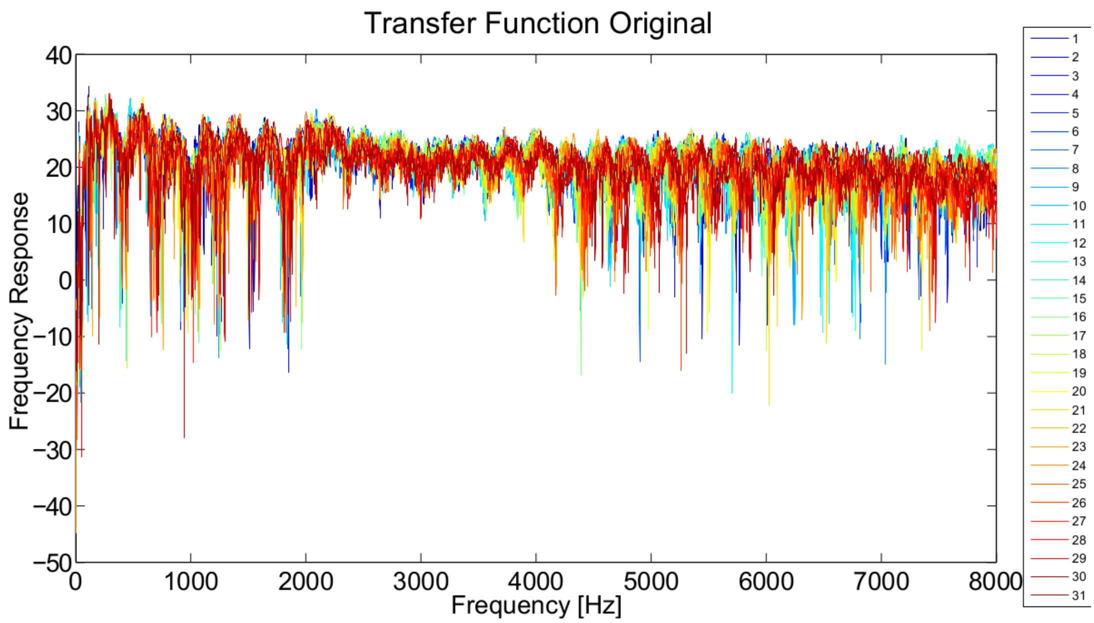
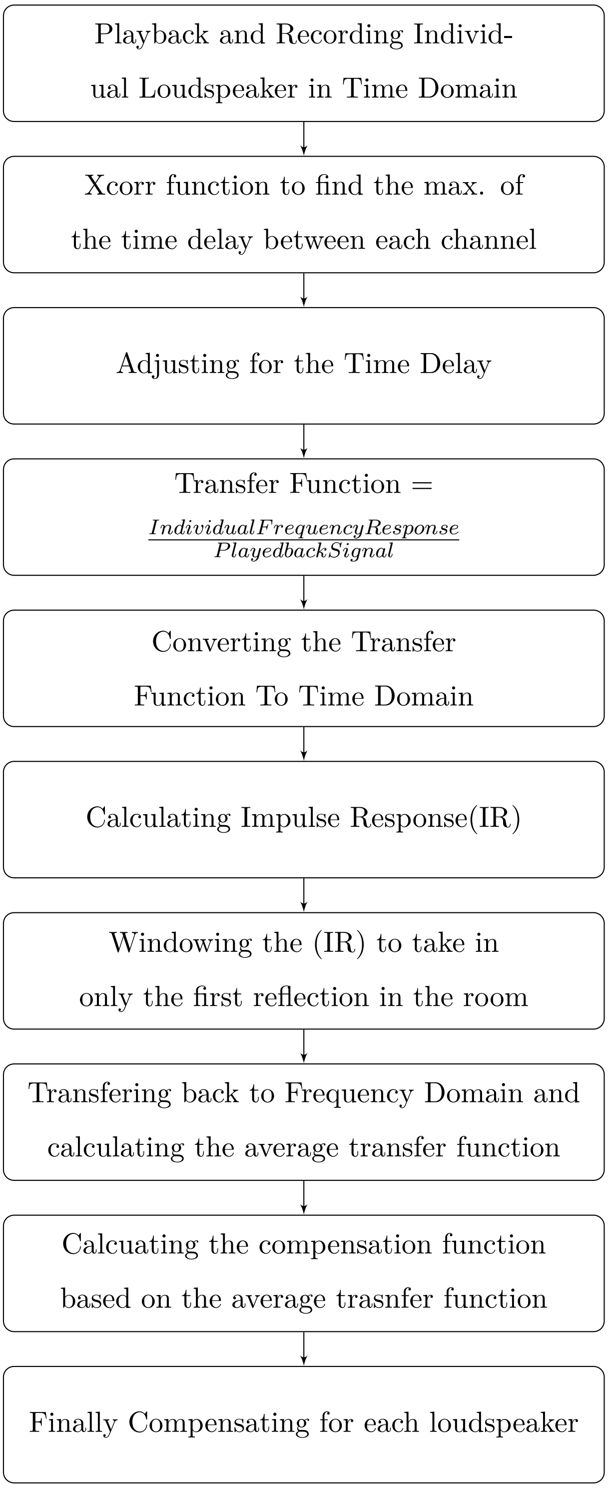


Figure 3.2: Original Transfer Function of the Array. The colors in the colorbar correspond to the 31 loudspeakers in the array. Room modes and reflections are also visible in the response.

The frequency range of interest was set to be 200-8000 Hz. This was chosen based on the fact that CI and NH individuals are most sensitive in this frequency range. Also calibration of the array was more accurate in this range. It is observed here in addition to the response from the loudspeaker, which ideally should be flat, the room modes are also excited.


Playback and Recording Individual Loudspeaker in Time Domain
Xcorr function to find the max. of the time delay between each channel
Adjusting for the Time Delay
Transfer Function = $\frac{IndividualFrequencyResponse}{PlayedbackSignal}$
Converting the Transfer Function To Time Domain
Calculating Impulse Response(IR)
Windowing the (IR) to take in only the first reflection in the room
Transfering back to Frequency Domain and calculating the average transfer function
Calcuating the compensation function based on the average trasnfer function
Finally Compensating for each loudspeaker

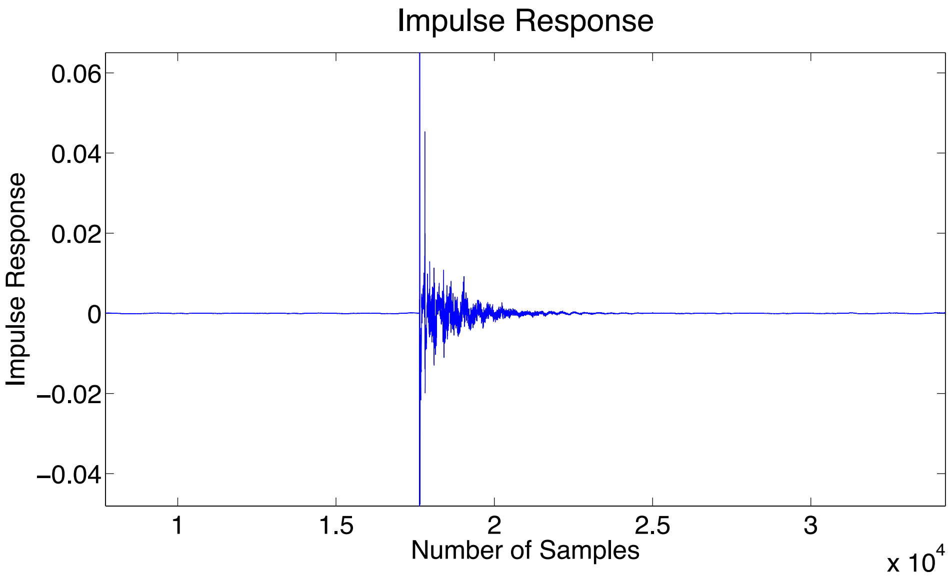


Figure 3.3: Original Impulse Response of a Loudspeaker.

The impulse response of the room is presented in figure 3.3. One can see that the second major peak occurs approximately at 296 Hz. This when calculated corresponds to the first reflection from the ground.

In order to study the response of the loudspeaker alone the impulse response was recovered just before the first major reflection.

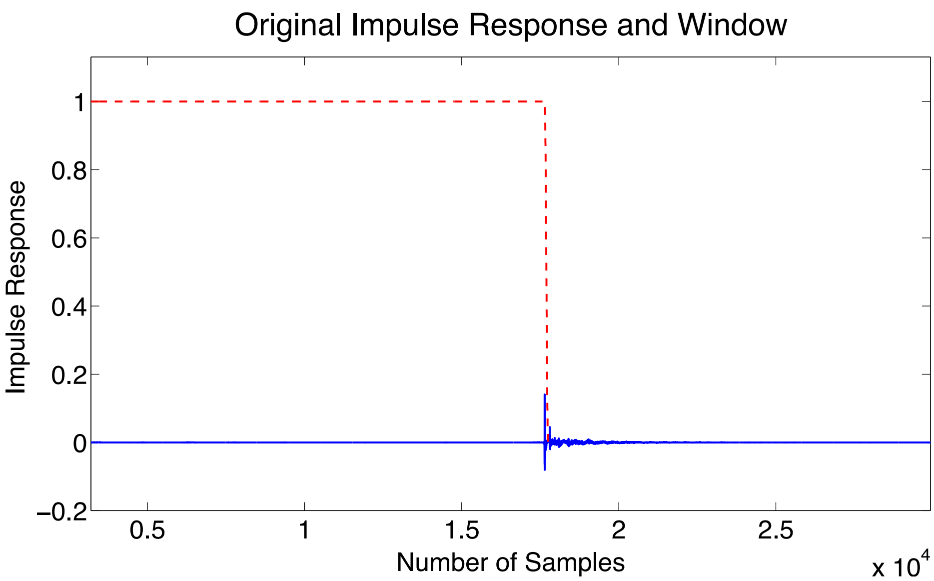


Figure 3.4: Window Implementation A window is applied to recover the response before the first reflection. The first reflection occurs due to the floor.

This was done by applying a hanning window as shown in figure 3.4. The resulting windowed impulse response is shown in figure 3.5 and it now consists of a peak mainly due to the direct wavefront.

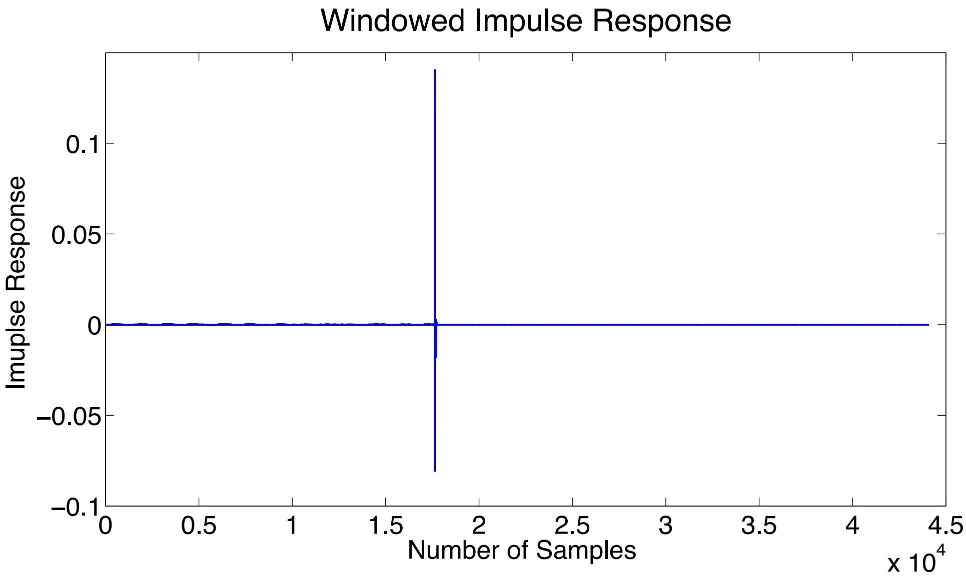


Figure 3.5: Response after filtering out the effect of reflection from the ground.

The window was also applied to the transfer function. It gave an approximation of the loudspeaker response after excluding the direct reflection from the ground.

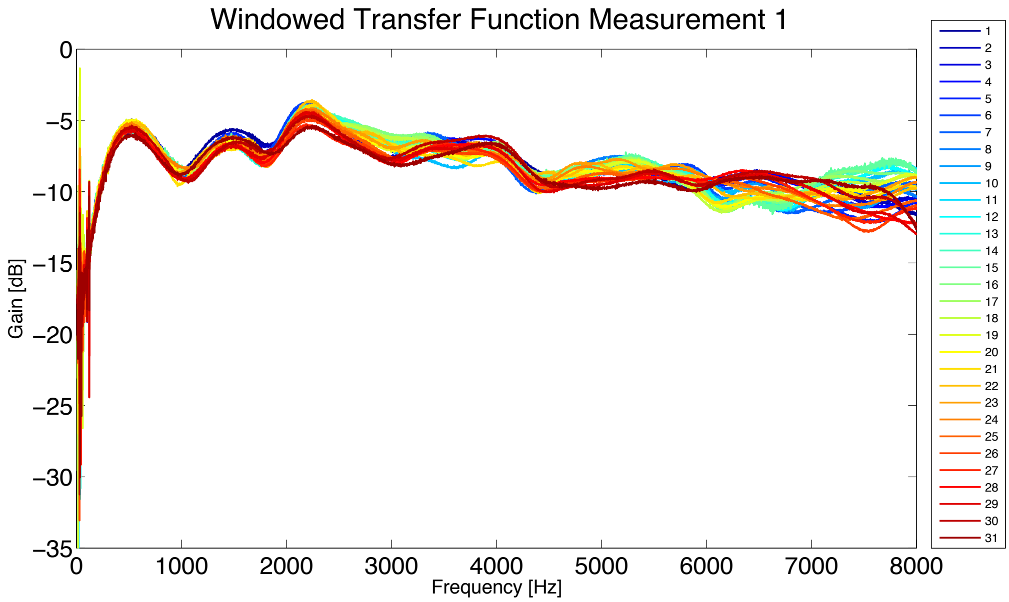


Figure 3.6: Windowed Transfer Function from 1st set of Measurement. The colors correspond to the 31 loudspeakers in the array.

On comparing figure 3.6 and 3.7 where the measurements were done on two different occasions it can be said that the array is stabilized enough to produce the same response for different measurements hence negligible discrepancy is expected in results from different trials.

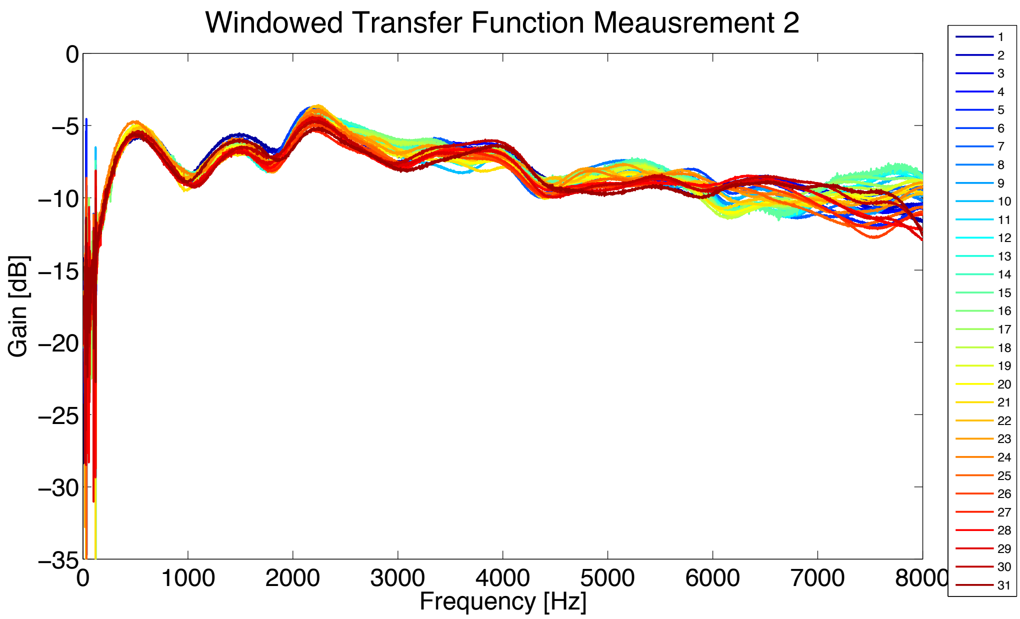


Figure 3.7: Windowed Transfer Function from 2nd set of Measurement. The colors correspond to the 31 loudspeakers in the array

The final step for the calibration was to compensate for the differences in the different loudspeakers. This was done by calculating the deviation in the response of each loudspeaker from the average transfer function of 31 speakers. This deviation was finally used to equalize the response from the different loudspeakers. The compensated transfer functions are presented in figures 3.8 and 3.9 respectively. A very close proximity is observed between the response of different speakers.

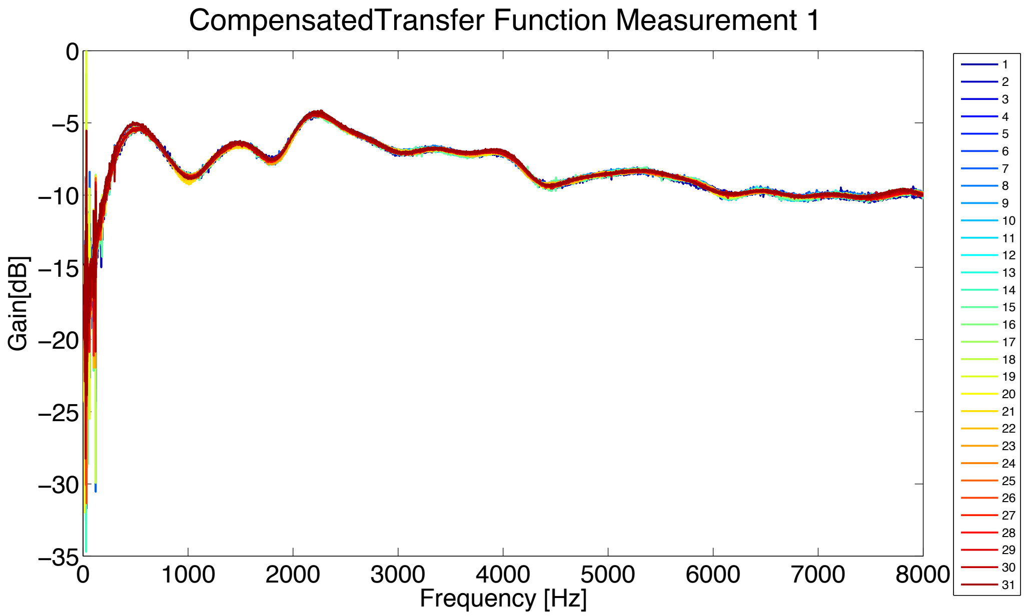


Figure 3.8: Transfer Function including compensation from 1st set of measurement. The colors correspond to the 31 loudspeakers in the array

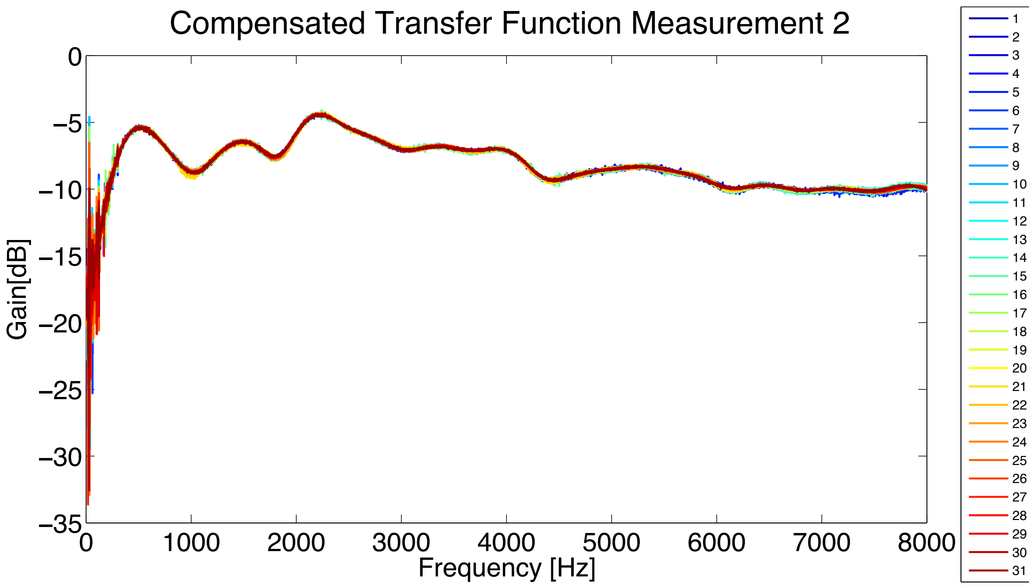


Figure 3.9: Transfer Function including compensation from 2nd set of measurement. The colors correspond to the 31 loudspeakers in the array

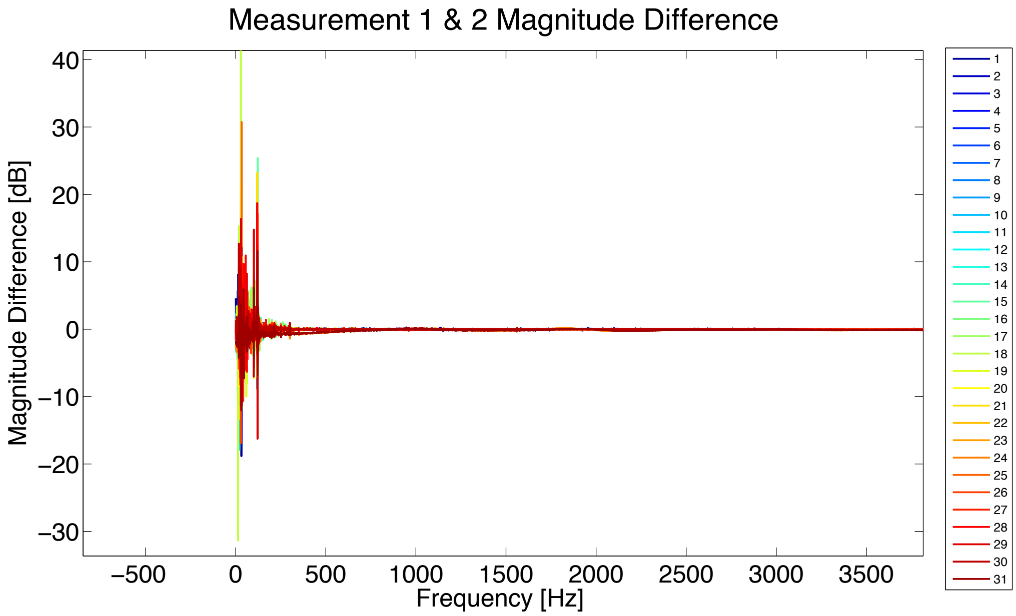


Figure 3.10: Transfer Function Magnitude Difference: The accuracy in the repeatability of the measurements is tested by measuring the response on two occasions and subsequently plotting the difference.

4 Auditory Source Width Dependant Signal Generation

Octave band Gaussian noise centered at 500 Hz and 2500 Hz were used to generate the signals with different auditory source widths. Each speaker of the array is separated by a distance of 6°. The whole array can be split in two equal halves i.e. the left half and the right half corresponding to $-90°$ for the 1^{st} speaker to $+90°$ for the 31^{st} speaker. In order to achieve the difference in the ASW a cosine squared window presented in equation 4.1 was used to get the desired signals. The sounds were centered at three different positions in the loudspeaker array. These are $\pm 30°$ and 0° i.e left right and center of the array. The ASW of the sounds were changed by changing the intensities of the signals being fed into the different loudspeakers. A pictorial representation of the auditory source width generation through the ring is presented in 4.1. For instance in order to generate a signal with auditory source width of 30° only the colored speakers will emit sound.

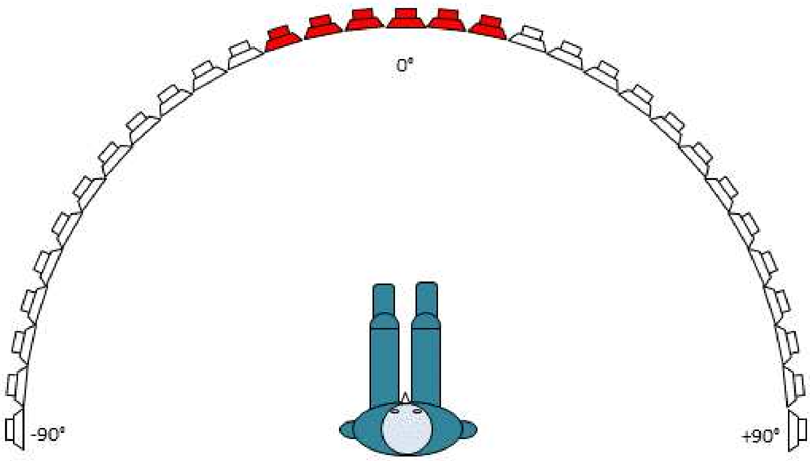


Figure 4.1: The Ring and ASW generation

$$a(S.P, n) = \sqrt{0.5 * [1 + \cos((\frac{2pi}{2W_{src}}) * min(W_{src}, |\theta(S.P) - \theta_{ch}(n)|))]} \tag{4.1}$$

W_{src} is half the ASW, $\theta(S.P)$ is the angle which defines the source position from the center of the ring. $\theta_{ch}(n)$ is the angle made by the n^{th} loudspeaker with the center. The 16^{th} loudspeaker is the central loudspeaker corresponding to a S.P(source position) of 0°. The function a(S.P,n) represents the intensity from each channel. In order to achieve the desired source width the intensities from each speakers are added. For measuring the ASW thresholds two reference and a single target signal were played to the participants. The 2 reference signals had the same ASW while the target signal was a different one. The expression $min(W_{src}, |\theta(S.P) - \theta_{ch}(n)|)$

makes sure that only the desired number of speakers emit noise for a particular auditory source width. For instance when the difference $|\theta(S.P) - \theta_{ch}(n)|$ becomes greater than W_{src} then what is left is $\cos\pi$ which is -1 and hence it results in a(S.P,n) = 0. Hence only the central speaker and 2 adjacent speakers on each side of the central one will be activated.

5 Interaural Time Difference and Interaural Coherence

In order to study the relation between auditory source width (ASW), interaural coherence(IC) and the interaural time difference (ITD) artificial head was used to record the sound samples. The artificial head was NEUMANN manufactured KU 100 i.e. having a binaural stereo microphone with frequency range of operation between 20Hz-20KHz. Two types of sound samples were used to determine which signal correlates better with the idea of ASW. These are -

- Independent Running Noise - The noise produced through each loudspeaker is different and it also varies each time the signal is played through the loudspeaker array. Two different broad band noises centered at 500 Hz and 2500 Hz were used.
- Equal Frozen Noise - A broadband noise centered at 500 Hz was used. In this type of signal generation the noise remains same for each loudspeaker.

After initial measurements independent running noise was chosen for the ASW threshold experiments. Sounds of different auditory source widths were played through the array and binaural recordings were made through the artificial head. Matlab function xcorr was used to calculate the interaural time differences (ITD) of the sounds based on the differences in their ASW.

5.1 ITD

Before measuring the interaural time difference(ITD) using the artificial head a theoretical value of ITD was calculated using the diameter of the artificial head which was 0.18m. The geometry of the ring and head is presented in figure 5.1 and 5.2. Considering a situation where the sound comes from a source position$(S.P) = 30°$. The diameter of the ring is 3.75m. The distance between the head center and the center of each speaker is 1.75 m.

The time difference was calculated as-

$$t_L = 1.88/340 = 0.0055s \tag{5.1}$$

$$t_R = 1.73/340 = 0.0051s \tag{5.2}$$

$$ITD = t_L - t_R = 0.4ms \tag{5.3}$$

Where t_L and t_R are the times taken by the sound to reach left and right ears respectively. The measured values of ITD are presented in figure 5.3

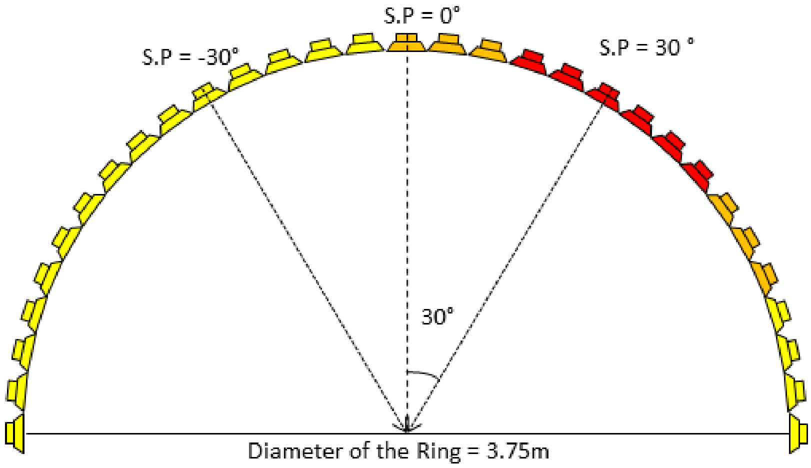


Figure 5.1: Ring Dimension

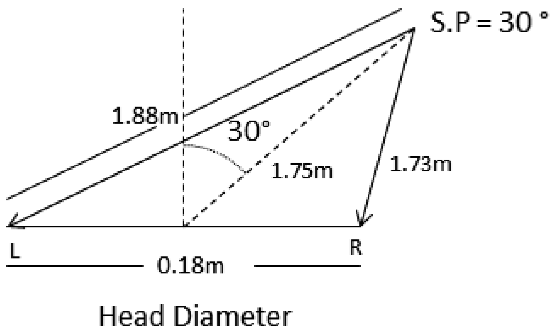


Figure 5.2: ITD Theoretical from Head Dimension

It is observed that the ITD for left and the right positions is very much around 0.4 ms for the smaller angles of ASW. However a slight difference in the ITD values appears as the ASW increases.

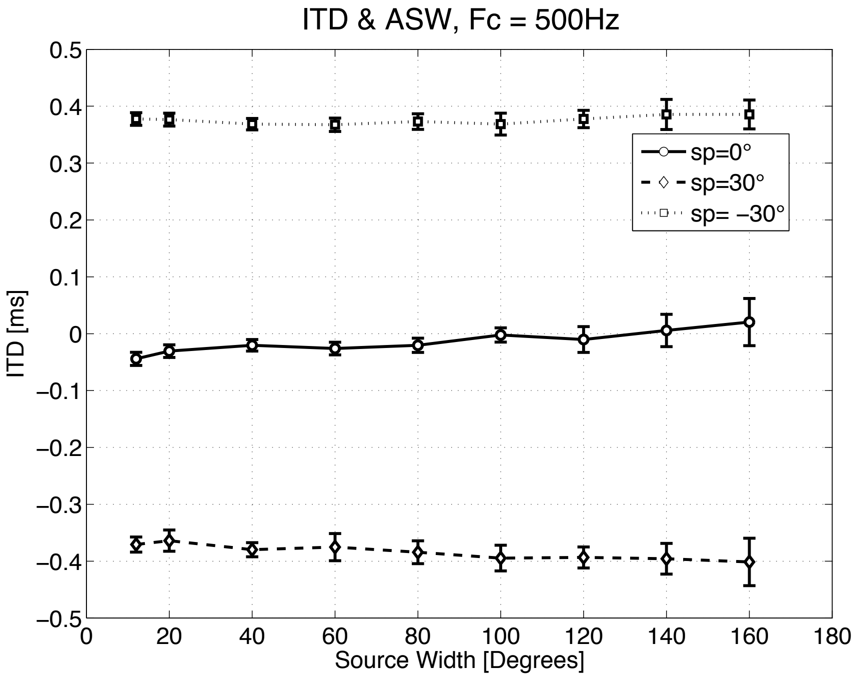


Figure 5.3: ITD and ASW comparison.

As a matter of fact a sound source lying directly in front of the listener should have an ITD value of 0s. However an oddity here is that the measured value of ITD for a source position of 0° i.e the central position is not zero. This can be attributed to the plausible reflections in the room and the asymmetry in the position of the ring in the room because the ring is not in the geometric center of the room. The results were obtained from averaging 20 recordings for each source position and auditory source widths. The results show some deviations in ITD's corresponding of larger ASW. A comparison of ITD values and the source position for a reference auditory source widths of $S_w = 40°$ and $S_w = 80°$ is again presented in figure 5.4. These are absolute values and the ITD's are presented in more conventional way. The ITD for the right side i.e.$S.P = +30°$ is slightly larger than for the left side. In addition the ITD corresponding to ASW of 80° is larger than for 40°.

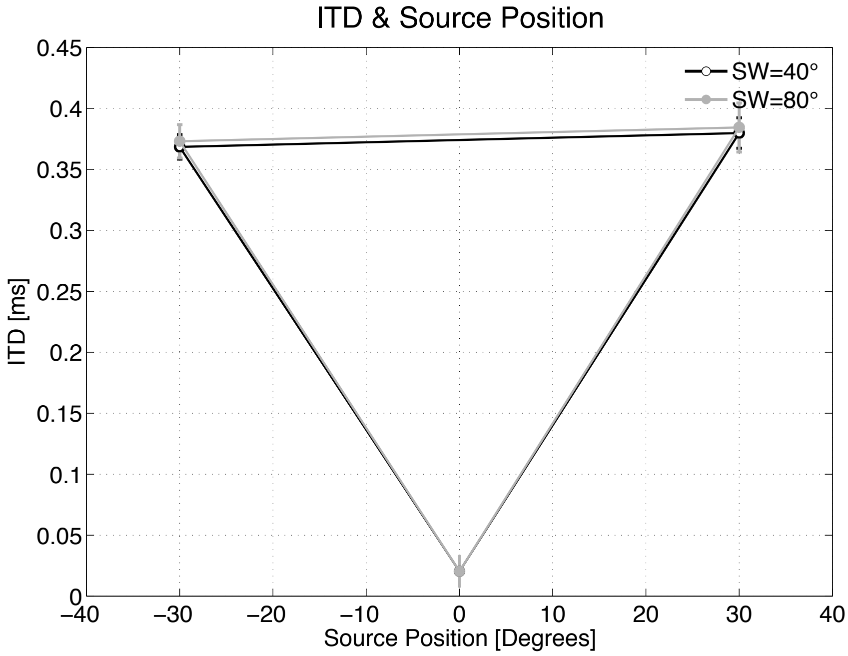


Figure 5.4: ITD and ASW comparison 2

5.2 IC

Interarual coherence(IC) is calculated by using the expression 5.4. $S_L(f)$ and $S_R(f)$ are the signals recorded with left and right microphones of the binaural head. IC helps us to decide how much to the left, right or center a sound is located. It is an interesting way of studying the perception of auditory source width. In chapter 2 under the literature subsection a detail description is presented on the relation between ASW and IC.

$$IC = \frac{| < S_L(f) * S_R^*(f) > |}{\sqrt{< S_L(f) * S_L^*(f) > * < S_R(f) * S_R^*(f) >}}, \text{[Fritz MENZER 2010]} \tag{5.4}$$

On observing figure 5.5 the premise of the inverse relation between ASW and IC as presented in the literature [Sato and Ando 2002] is repeated with the binaural recordings done with artificial head in this thesis work. However the corresponding IC value for a ASW of 12° which was expected to be maximum of the measured IC values is less than that for 20°. No plausible reason was found for such a behavior hence for the ASW experiments with participants the minimum value of ASW was restricted to 40°.

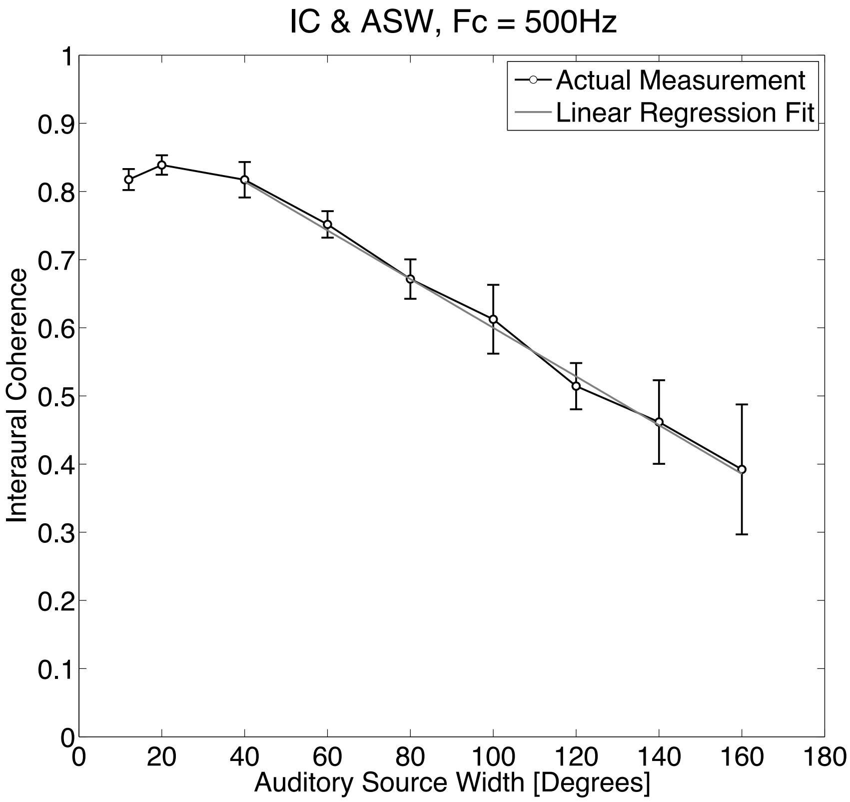


Figure 5.5: Relation between auditory source width and interarual coherence calculated with the help of binuaral recordings obtained using an artificial head. Black curve represents the result from the actual recordings while the gray plot refers to the extrapolated curve obtained through linear regression.

In figure 5.6 a relation between the calculated IC value for two different noise is presented. The square symbol plot corresponds to frozen noise at 500Hz. It is seen that the IC value from source width of 40°· almost becomes unity i.e. it is independent of the auditory source width. This is attributed to the fact that frozen noise is highly correlated. Contrary to the frozen noise the interaural coherence value for independent running noise at 500Hz depicts a fine inverse relation with the auditory source width. However for a center frequency of 2500Hz independent running noise shows though vaguely inverse yet a very staggered relation with auditory source width.

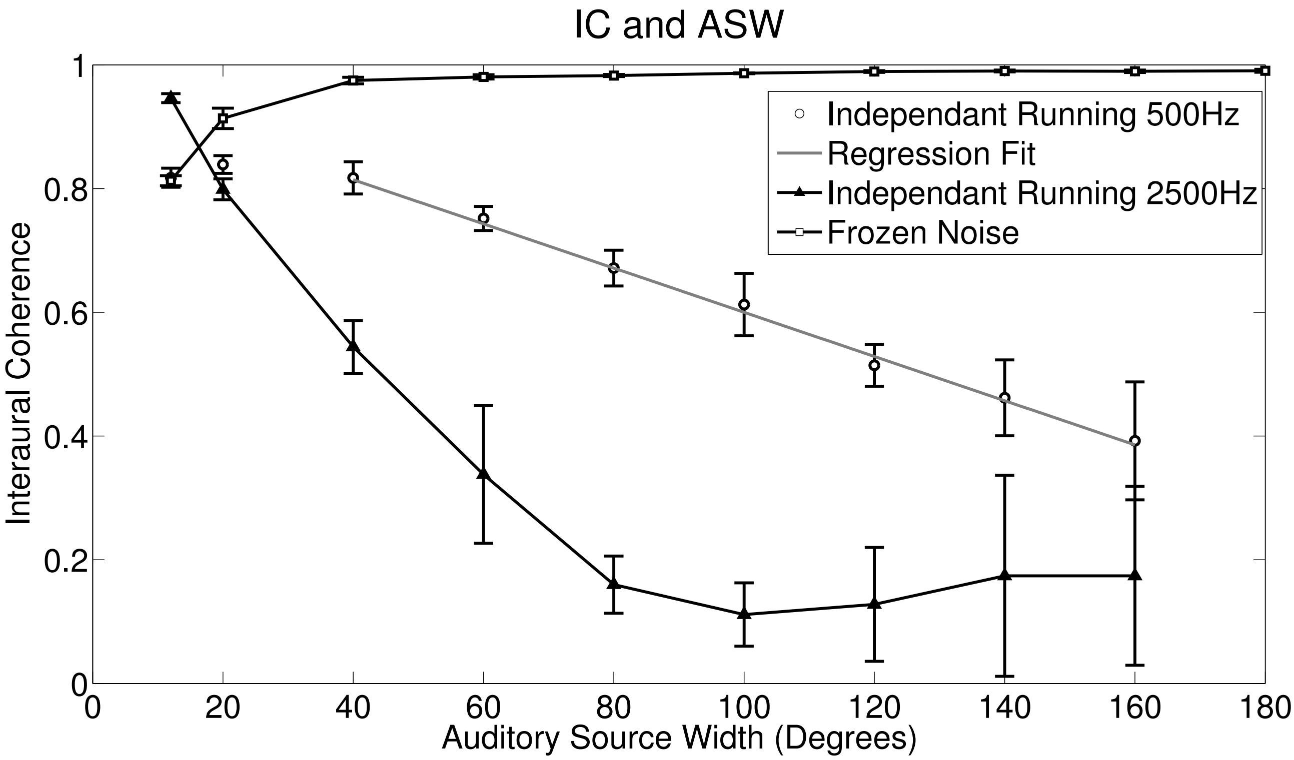


Figure 5.6: Relation between auditory source width and interarual coherence calculated for different noise types at different frequencies.

6 ASW Threshold Experiments

Auditory source (ASW) thresholds were measured using 3 alternate forced choice procedure(3AFC). Each experiment consisted of different runs and for each run the participants were presented with three noise signals. The details of signal generation is presented in chapter 4. Out of the three stimuli two were the same while one of the signal differed in terms of the auditory source width. To say it in another words there were two reference signals and one test signal. The reference signals were identical signals having the same ASW. In each presentation three stimuli were presented to the participant. They were played back to back with a time difference of one to half a second between each noise stimuli. The participant had to chose one of the three stimuli and hence 3AFC. The threshold measured presented in the difference of the ASW value between the reference signal and the test signal. Two types of measurement methods were used. These are-

6.1 Adaptive Stimulus Procedure for NH Participants

In this method the succeeding stimulus depended upon the response to the preceding stimulus. Since it was expected that NH participants would be better in measuring ASW thresholds hence an adaptive stimulus procedure was used in these experiments while constant stimulus procedure was followed for the experiments with CI. The method followed to measure the thresholds was Transformed 1 up and 2 down method [H.Levitt 1970]. Fundamentally the threshold was increased if there was one wrong response and the threshold was decreased after two right responses. The two different noises were centered at 500 Hz and 2500 Hz. While for a stimuli of 500 Hz the reference source width was varied from 40° to 120° in a step size of 20° whereas for a stimuli of 2500Hz only reference source width of 20° and 40° was measured. Thresholds for three different source positions(S.P) were measured. These were left, center and right corresponding to the S.P of $-30°, 0°$ and $+30°$ respectively. For each reference source width three repetitions were carried out and the presentations were randomized. The mean value of the thresholds is presented in the next section under results.

6.2 Constant Stimulus Procedure for CI Participants

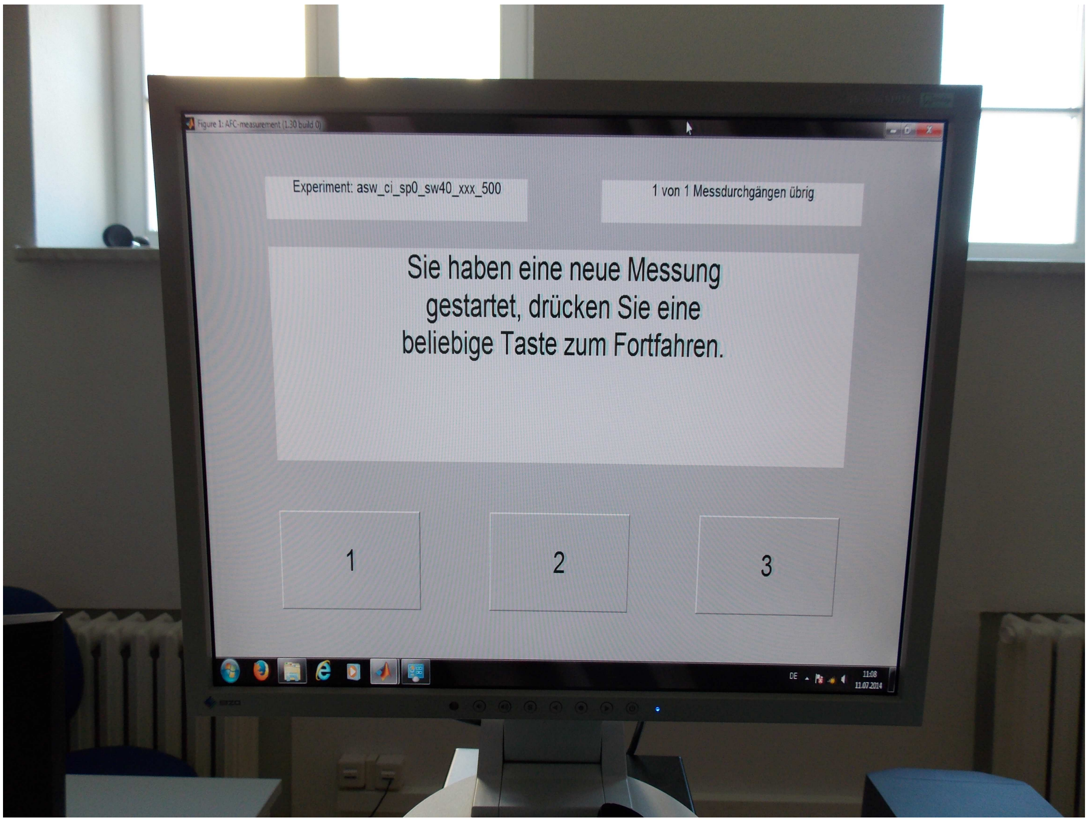


Figure 6.1: Screen-shot of the display for CI participants.

The threshold measurements for CI participants were performed using constant stimuli procedure. In this method an estimated range of thresholds, source width in the present work, were presented to the participant multiple times in a randomized fashion. The experiments were restricted to the source position of 0° only. Center frequency of only 500 Hz was used for these measurements. It was expected that the performance of the CI's may differ significantly owing to difference in their hearing loss. Thus the first experiment with the CI consisted of a stimuli with a reference source width of 40° and the target stimuli had source widths of 80°, 120° and 160°. After the first experiment the performance of the participant was observed in order to check the correlation of the results in respect to the chance level performance i.e. 33.3 % for a 1 up 2 down 3AFC procedure. If it was above the chance level only then the second experiment was proceeded where reference stimuli had the same source width of 40° while the target stimuli had the source width of 60° and 100°.

However if the participant was not able to perform above chance level then the second experiment was skipped and a third experiment was carried out with a stimuli having a reference source width of 80° while the target stimuli had a source width of 120° and 160°. The results are presented in the next section.

The candidates for the ASW thresholds experiments were people who have been implanted with the prosthesis at the department of Universitätsklinik für Hals-, Nasen- und Ohrenheilkunde, Universitätklinikum, Magdeburg, Germany. A total of four CI took part in the experiments. They had experience with listening tasks as they have been taking part in various listening experiments for different studies at the department.

Figure 6.2: A CI performing the experiment.

Before the actual measurements a small test session was ran to know if the CI were comfortable with sound pressure level of the stimuli. Upon the stimuli presentation the CI were asked to choose which sound did they perceive as different out of the 3 sounds at each presentation. Instead of pressing their choices themselves they were asked to speak out their preference while an instructor present in the laboratory entered their choices through keyboard.
All the CI were only German speaking participants therefore the message displayed on the screen was in German language. A screen-shot of the display and a measurement session is presented in figure 6.1 and 6.2 respectively.

7 Results

7.1 Normal Hearing

An invitation was sent out to the university students to take part in a study pertaining to listening experiments. A total of 12 students took part in the study with 7 males and 5 females. Every one confirmed that they haven't had any kind of hearing impairment in their life. Each participant visited the free field room on two different days and completed the listening task in total of four sessions. Before the start of the experiments there was a demonstration measurement having a limited number of repetitions in order to get the participants acquainted with the task.
In the first and third experiments the stimuli was centered at a source position(S.P) of 0° i.e the center of the ring. Two different sessions with stimuli having center frequencies of 500Hz and 2500Hz were conducted. Three stimuli were presented at a time and the participants were asked to choose the broadest of the sounds. They were provided with a keyboard and asked to press 1,2 or 3 according to record their preferred stimulus. The results are presented in figure 7.1.

The aspect worth mentioning here is the response of the participants to a stimuli having a reference auditory source width of 120°. Almost 9 participants were not able to differentiate between the reference and the target stimuli for a reference auditory source width of 120°. Since the reference and the target stimuli had ASW >= 120° therefore it can be said that it was too wide for most of the participants to acknowledge any difference between the two. Hence for the participants namely NH 1,2,4,5,6,7,8,10 and 11 the error bars are not plotted. Every time it occurred the screen displayed the message "The track has been skipped". On observing the panel NH 11 it can be said that the participant was not able to perceive a difference in the stimuli even for a reference source width of 60°, 100° and 120° but was able to distinguish the target stimuli for a reference source width of 80°.

The thresholds for the stimuli centered at 2500Hz have a significant difference between the reference source widths of 20° and 40°. The threshold for 40° is considerably higher than for 20°. On comparing the responses for stimuli having center frequencies at 500Hz and 2500Hz it can be said that the response for 500Hz is relatively flat than the response for 2500Hz. Furthermore the standard deviation from the mean in case of 2500Hz is quiet greater than for the 500Hz responses.

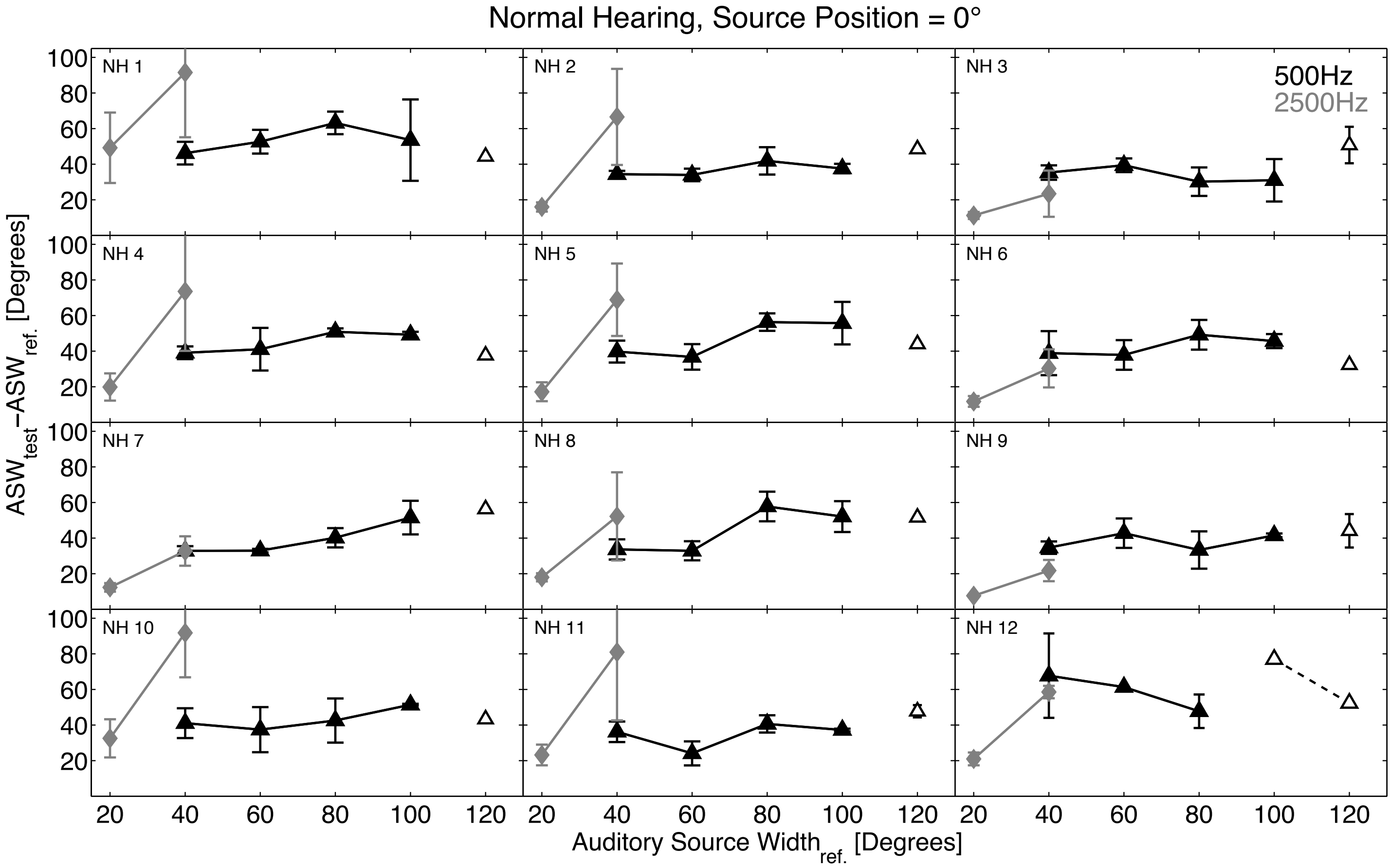


Figure 7.1: ASW thresholds as a function of reference auditory source width. Each panel represents the performance of an individual NH participant. Black and Gray plots correspond responses to a stimuli of 500Hz and 2500Hz respectively. Open black triangle symbols for ASW= 120° refer to 500Hz stimuli that was skipped by majority of the participants.

After the above mentioned experiments the remaining two sessions were conducted to observe the effect of source location on the ASW thresholds. It is important here to mention that there was a limitation in terms of the number of hours and visits each participant was willing to devote to the experiments. Therefore for a stimuli of 2500Hz only the source position of 30° with a reference source width of 40° was tested. For the stimuli of 500 Hz both left and right source positions with the same reference source width of 40° was measured. The results are presented in figure 7.2.

It is interesting to highlight that for a stimuli of 500Hz while participant NH1 is not able to perceive a difference when the source is positioned at $-30°$ whereas NH2,NH3,NH8 and NH12 are not able to detect the difference for the source centered at 30°. Furthermore participants NH5 and NH11 are not able to perceive the difference on both of the sides. As a result one can say that the performance within participants differ more when the sound source is positioned on the sides as compared to the center.

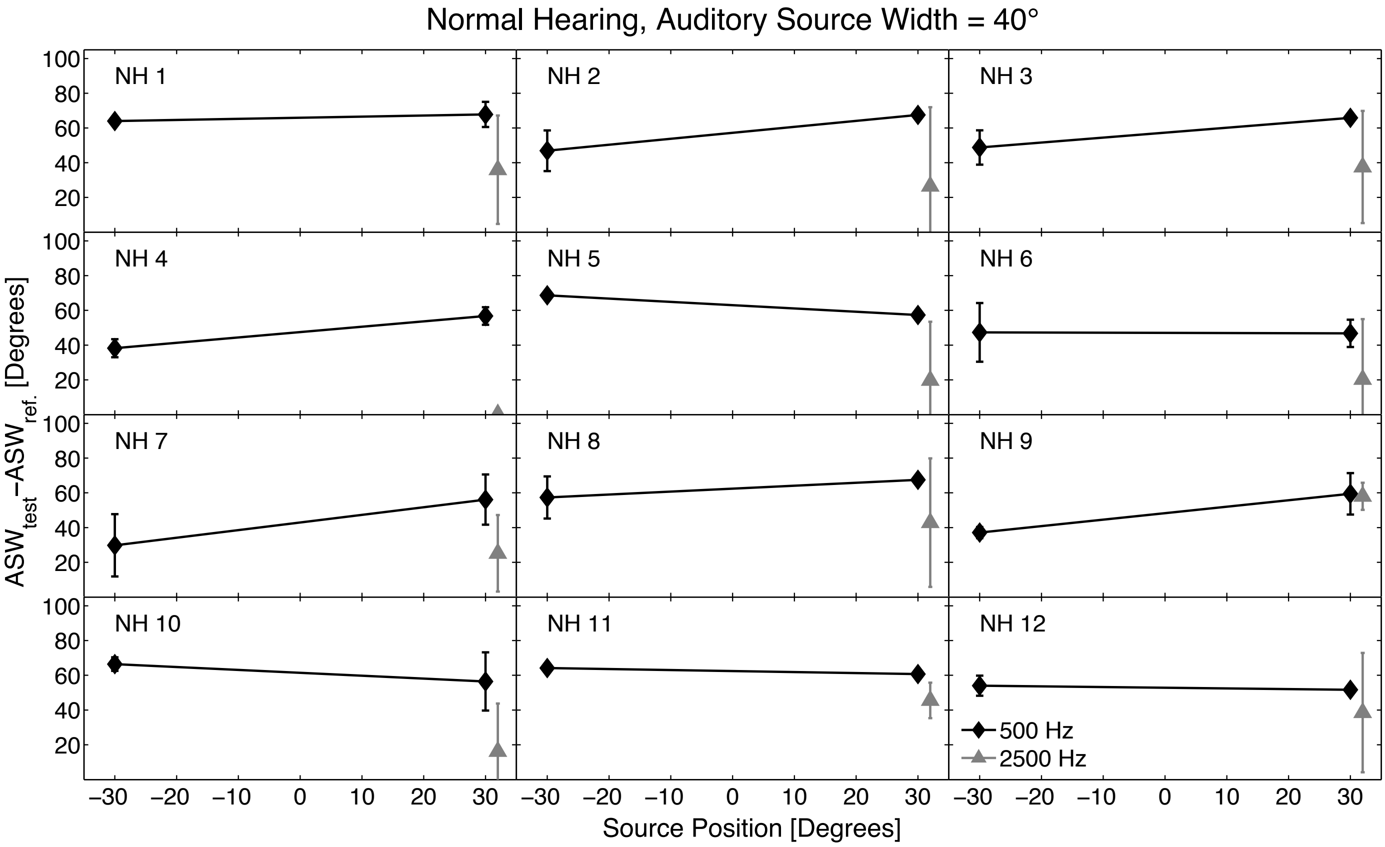


Figure 7.2: ASW threshold as a function of source position for an ASW of 40°. Each panel represents performance of an individual NH participant. Black and Gray plots correspond to the stimuli of 500Hz and 2500HZ respectively. The 2500 Hz stimuli actually orginated from a source position of 30° however to show it more clearly in the figure it is shifted to 32°.

Figures 7.3 and 7.4 present the mean ASW thresholds. While 7.3 corresponds to the results for a stimuli positioned at the center of the ring whereas 7.4 refers to the mean performance for a stimuli centered on left and right sides. On examining the two figures it is apparent that the threshold is higher when the source is centered on the sides compared to when it is positioned at the front.

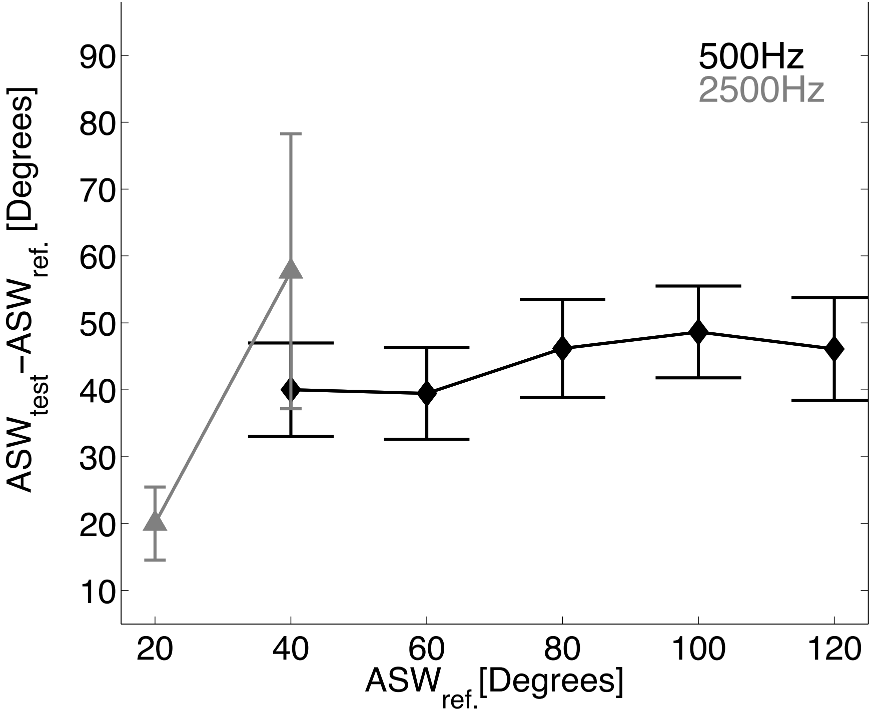


Figure 7.3: The black and gray curves represents mean threshold values for a stimuli of 500Hz and 2500Hz respectively for a source positioned at 0°.

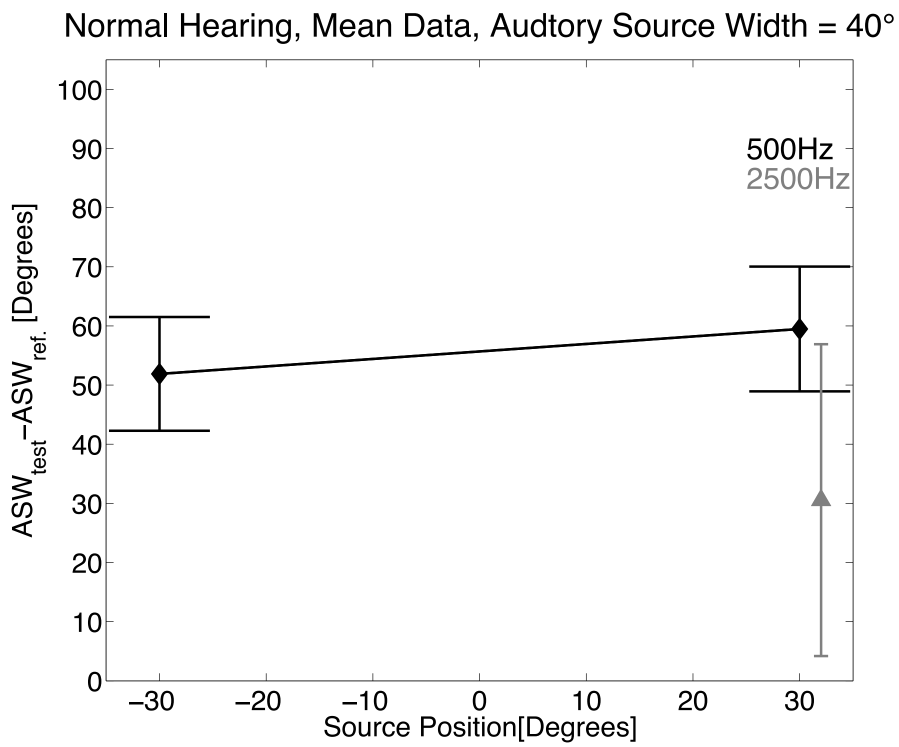


Figure 7.4: The black and gray curves represents mean threshold values for a stimuli of 500Hz and 2500Hz respectively having an ASW = 40°.

7.2 Cochlear Implantees

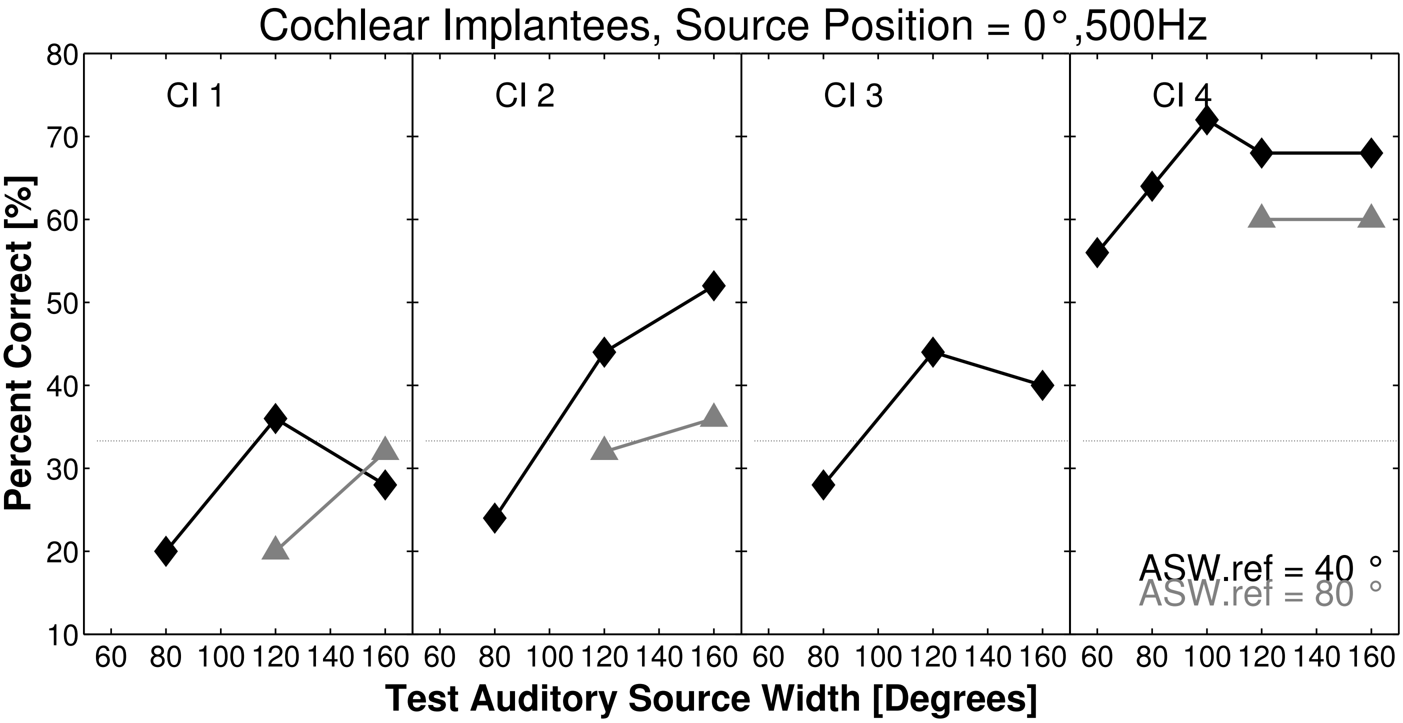


Figure 7.5: ASW thresholds for CI participants. The black and gray curves refer to the responses for stimuli of ASW equal to 40° and 60°. The dashed line corresponds to the minimum possible percent correct score for a 1 up 2 down procedure which is equivalent to 33.3 %. One of the participant CI3 was not able to perceive a difference for the reference source width of 30°.

The first three participants namely CI1, CI2 and CI3 were clearly not able to perceive difference for the test source width of 80° because their performance is well below the chance level. The performance of all the above mentioned participants for a source width of 120° was above chance level i.e. above 33.3 % percent correct. Participant CI1 aged 68 years had a unilateral implant compared to rest of the participants who had a bilateral implant. CI1 was implanted with the prosthesis on his right side in 2013.

CI2 aged 50 years got his prosthesis on the right and the left side in the years 2010 and 2011 respectively. CI3 aged 46 years got both of his years implanted in 2013. CI4 aged 62 year got bilaterally implanted in 2002, right ear and in 2005 in the left year.

It can be distinguishably observed that CI4 has been an outstanding performer. It was even possible for CI4 to measure the thresholds for an additional source width of 60° and 100°. The performance lies between 68-74% correct response. It is worth to mention here that CI4 was the participant who have had prosthesis for the longest time as compared to the other participants. Hence some of his performance can be attributed to the fact he could be the participant who had the maximum time to learn and adapt to the hearing device. Interestingly he is also the participant who performed with a constant 60 % correct response for a stimuli with reference source width of 80° and irrespective of the target source width.

8 Discussion

The auditory source width thresholds for NH participants for a stimuli of 500Hz with source position at 0° are transferred to interaural coherence(IC) domain. It is done by tracing the corresponding IC values for respective ASW calculated with the help of artificial head, figure 5.5. The individual and the mean IC values are presented in figure 8.1.

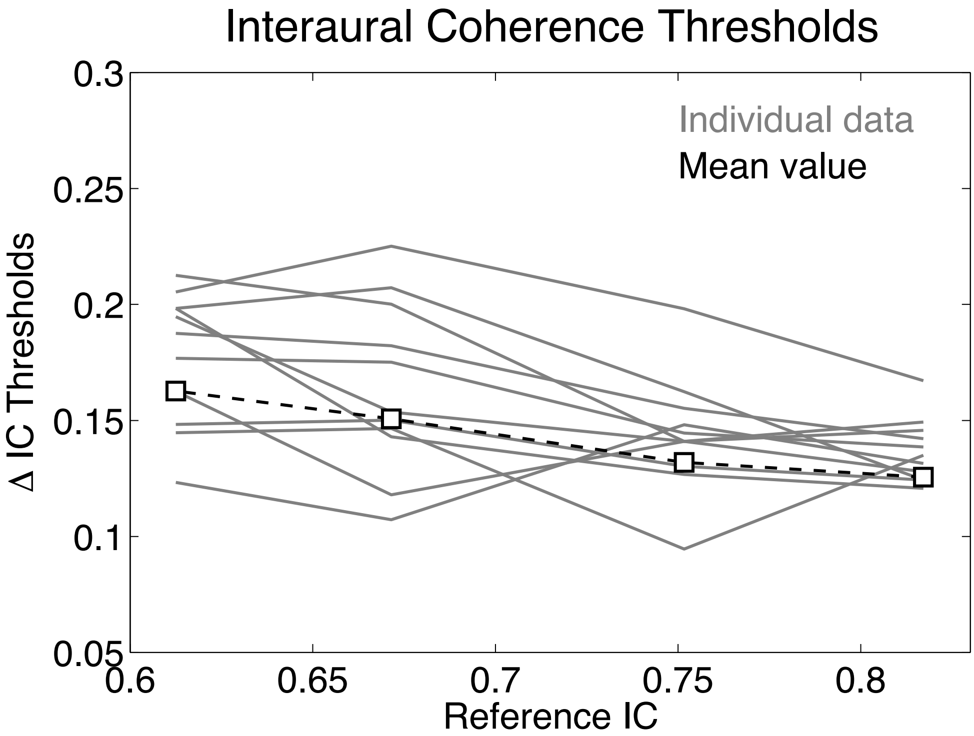


Figure 8.1: The figure presents IC thresholds as a function of reference IC calculated using the binaural recordings done by the artificial head.

The mean threshold value decreases with the increase in reference IC. This can be accounted to the fact that as the IC increases the sound is perceived to be coming more and more directly from the front. Ergo it is evident that for the sound source located in the front will have lower IC thresholds compared to the sound source located at the sides.
If individual data is inspected it can be noticed that for nearly half of the NH participants threshold for a reference of 0.65 was greater than for IC = 0.6 .

It is appropriate here to mention that when we actually commenced on the thesis it was decided to be one of the pursuits to present a reasonable comparison between the headphone based apparent ASW studies by [Whitmer, Seeber and Akeroyd 2012] and the free field experiments in the present study. The results of the headphone study coincides precisely with the results from one of the earliest studies on IC by [Pollack and Trittipoe 1959], gray plot in the figure 8.2.

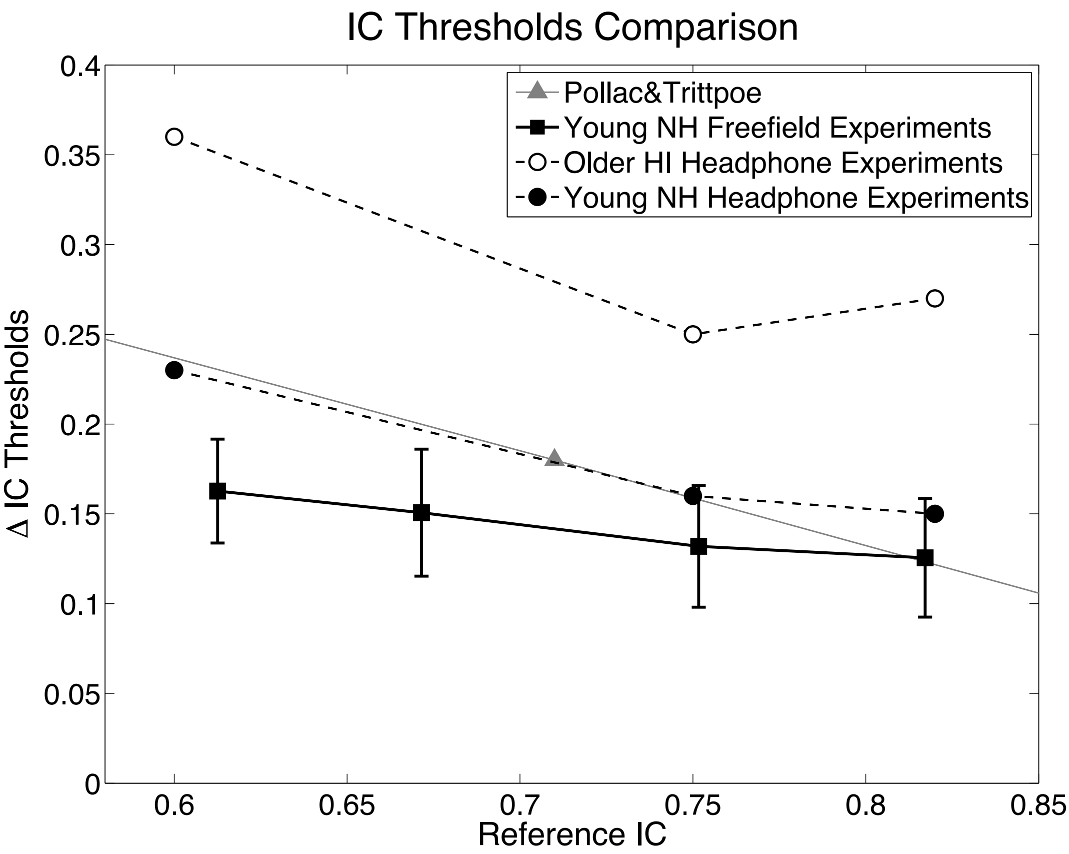


Figure 8.2: The figure compares the mean IC thresholds for headphone experiments(literature) and free field measurements(thesis work).

A detailed description of the headphone experiments can be found in the Literature chapter. Resuming the above discussion and combining the results from two studies i.e. figure 2.14 and 8.1 a comparison between IC thresholds is presented in figure 8.2.

The older hearing-impaired(HI) participants belonged to the age group of 46-75 years and the younger normal hearing(NH) participants in both the experiments were below 40 years of age. When [Whitmer, Seeber and Akeroyd 2012] performed their listening experiments none of the older HI participants used any sort of hearing aid device. Upon observing the comparison it is clear here that older HI participants indeed are significantly challenged while detecting IC thresholds.

The IC thresholds for young NH participants from the freefield measurements and the headphone representations differ by a factor of 1.6 for the smallest reference IC. However this difference reduces substantially for the highest measured reference value of 0.82.

A conjecture at the moment for the difference in the thresholds between literature and present study is the fact while in the headphone experiments the stimuli were based on different IC values. Whereas in the free field studies ASW was the distinguishing factor in the stimuli but not the IC. Hence IC could account for the difference in the thresholds for the two studies. It is essential here to remind that the presents study engages ASW in realistic environment compared to the headphone based playback signals. Furthermore it consolidates one very important fact here. Interaural coherence(IC) plays a crucial role in perceiving the auditory source width of a sound(ASW).

The present study with CI is first of its kind where ASW perception by CI is investigated for the first time. However it is interesting to relate the present work with other studies investigating sound localization in CI. For example [Zheng et Al] demonstrated that CI have higher difficulty in localizing a signal in presence of noise and reverberation. They concluded that NH listeners are not effected by reverberation while CI face serious difficulty. In literature ASW is discussed to be effected by a sound arriving in the first 60ms. Hence it can be said that ASW should not be really effected by the reverberation in the space. However the results from [Zheng et Al] establish a fact that reverberation does effect CI if not NH listeners. Taking a step further it can be said that while measuring the ASW thresholds for CI it could have been possible that the reverberation of the room played a significant role in the task in spite of measuring in a free field environment.

In a separate study [Kerber and Seeber] found for sound localization binaural ability of the person plays a significant role. If it is just the speech or conversation with a single person then it is fine with monoaural listening. However the moment situation has multiple sound or speech source the listener has to be dependent on binaural hearing. It is a worthy observation here that binaural listening plays a decisive role in perception of ASW. Therefor irrespective of the signal being speech or noise it in order to successively distinguish the difference in ASW binaural listening is essential.

9 Conclusion

Spatial perception of sound is influenced by numerous factors which provide relevant cues for processing auditory information. Auditory source width perception is a measure which provides valuable information on how one assesses the sound and space associated with it. Presently a limited researchers have addressed ASW perception in people although a substantial number of acousticians and academicians have investigated ASW in concert halls in order to criticize the sound quality of the space. However there is a need to realize the worth of studying ASW in hearing impaired people.

Only one such work is available in literature however it is limited to those hearing impaired individuals who don't have cochlear implants. CI have a serious difficulty in differentiating sounds from various sources and ASW does play a significant role in sound discrimination. After the threshold experiments it has been evident that ASW perception is relatively minute in Cochlear implantees as compared to normal hearing listeners. However there is a possibility that ASW perception in CI improves with the time as in the case of CI4.

The next stage will be to measure ASW thresholds in the presence of a masker. Since ASW perception can assist an individual in differentiating various sound sources hence it will be interesting to study ASW perception by CI in presence of a noise and secondly in presence of speech signal.

One way to expect improvement in the performance of the CI when it comes to spatial perception of sound is to include fine structure in the implant itself to a larger extant than what is possible today.

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